what's new

ISSUE 01

TELOS OMNIA AXIA

C A T A L O G   S U P P L E M E N T   2 0 1 0

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AUDIO | NETWORKS

TELOS

what's new! / telos / VX

TELOS VX: THE CONVERGENCE OF IP-AUDIO AND STUDIO TELEPHONE SYSTEMS

Telos VX is the world's first multi-studio VoIP phone system for broadcast. It's scalable to provide its advanced phone interface capabilities to facilities with a few studios or dozens.

VX uses Ethernet as its network backbone, a powerful yet simple way to share phone lines among studios and connect system components. VX plugs right into Axia IP-Audio networks, connecting multiple channels of audio and control via a single Ethernet RJ-45. If you don't have an IP-Audio network yet, VX works with Axia nodes to provide analog or AES audio and GPIO connections.

The VX uses standard SIP (Session Initiation Protocol) and RTP (Real-time Transport Protocol) protocols, so it works with your VoIP-based PBX or SIP-trunking Telco service to direct up to 80 lines of phone traffic. Gateways allow the VX to connect to traditional Telco lines, including T1/E1, ISDN, and POTS.

Call processing is sophisticated and flexible. Lines can be easily shared between studios. A web interface lets you assign lines to 'Shows', which can then be selected by users on the studio controllers. The call screening interface uses familiar Telos Status Symbol icons to let talent and producers know, at a glance, the status of each line. Each studio has its own dedicated Program-on-Hold input.

VX also features our most advanced audio processing ever. Each incoming line has its own individual hybrid, allowing multiple calls to be conferenced and aired simultaneously with noticeably superior clarity and fidelity. Each call may be assigned to its own fader for individual control of volume and on/off status during conferencing. AGC, adaptive Digital Dynamic EQ and call ducking are part of the VX toolkit, along with a breakthrough Acoustic Echo Cancellation (AEC) capability that solves the longstanding problem of feedback and echo in open-speaker situations.

The VX Producer PC application includes an integrated recorder, player, and editor. A producer may easily record and edit caller comments and send them to the studio for on-air play. In-studio DJ-style call recording, editing, and play is also simplified.

Smooth integration with studio mixing consoles is achieved via their common network connection.

The Telos VX system includes:
• **VX Engine** – Call control and audio processing engine
• **VX Director** – IP-based call controller
• **VX Producer** – Windows based software for call screening, recording and editing

VX ENGINE

The VX Engine is the heart of the system. This is where all hybrids are housed, call control and audio processing occurs. Since VX is an IP-based system, I/O is only limited by network bandwidth and throughput resulting in an unprecedented capacity of up to 80 available incoming lines, each with its own third-generation Telos Adaptive Digital Hybrid featuring Advanced Echo Cancellation developed by Fraunhofer. Up to 20 studios can connect to a single VX Engine, each with its own VX Directors and VX Producer call screening software for control. Talent can customize their telephone workflow with Show Profiles to store commonly-used show configurations – for talk shows, interview segments, or anything else – and recall them for instant use.

VX is equipped with a rich toolbox to help make caller audio sound its very best, no matter what kind of line or phone the caller is using. Smart AGC, noise gating, a high-pass hum filter, Telos' famous Digital Dynamic EQ, and a three-band adaptive spectral processor are all brought to bear on caller audio, while send audio gets its own sweetening which includes a frequency shifter, AGC/limiter and FhG's Advanced Echo Cancellation.

Remote control and configuration? Of course. VX's web-based configuration lets engineers work with VX from any networked location using a standard Internet browser.

And since VX employs VoIP technology, talent, producers and board ops have a wide range of phone, console and software interfaces at their disposal. There's VX Director, with its familiar phone-set design and big, bright color LCD display. VX Producer call-screening software, with a built-in soft-phone, turns any PC into a phone station. And direct console integration with Axia IP-Audio consoles that let talent make and take calls directly from the board. Each VX Engine supports up to 100 control devices, in any combination.

At only 2 RU in height, and 15” deep, the VX Engine can be neatly tucked away in any equipment rack.

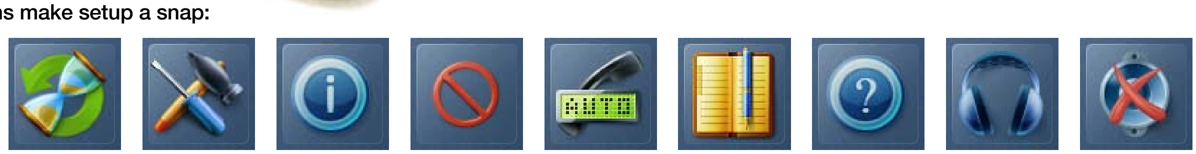
VX DIRECTOR

The new Telos VX Director phone is beautifully designed, with an attractive, friendly LCD color display that uses exclusive Status Symbols to let talent know what's going on in an instant. VX Director can handle up to 12 phone lines, providing detailed line status, caller information and fader assignments at a glance. The info-rich display shows caller ID for each line, along with time ringing-in or on-hold, and even screener comments from the VX Producer software application.

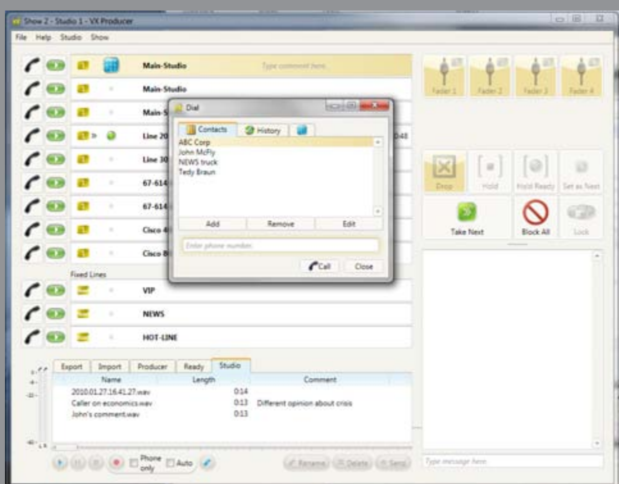
VX Director gives talent unprecedented flexibility. You can map groups of lines to a single fader, making it simple to take a queue of calls to air sequentially — or use the Fader Assign function to map lines to individual console faders for precision control of multi-call interviews or conferences. One-touch controls let talent step through queued calls, busy incoming lines, lock calls on-air, even start an external recording device.

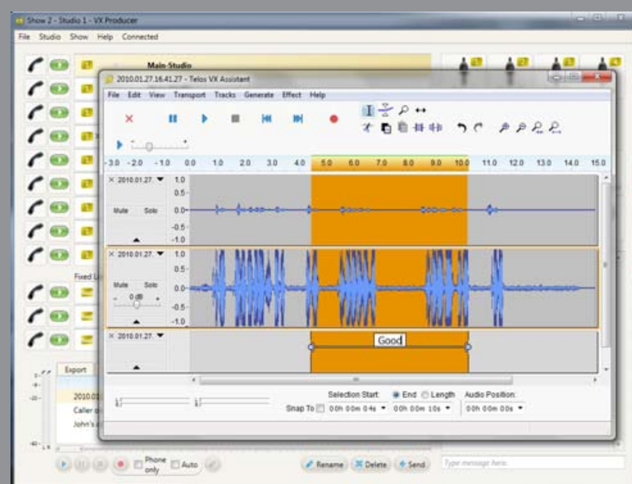
A built-in address book and call history log round out VX Director's features; as with the rest of the VX system, each VX Director has its own web server for easy remote configuration and software upgrades.

Intuitive icons make setup a snap:



History | Setup | Status | Block All | Auto Answer | Address Book | Status | Mode & Select Studio | Mute





VX PRODUCER

Windows™-based VX Producer software streamlines call screening, recording and editing with its intuitive, user friendly interface. VX Producer gives talk show producers all the information they need, including caller status, screener's comments, Caller ID and time per call for up to 12 lines. Like the VX Director, VX Producer's Fader Assign feature lets you map calls to up to eight specific console fader locations if you wish, and includes an address book and call history log. There's also a Chat function that lets producers and talent communicate quickly off-air using VX Producer.

But VX Producer is much more than just a call screening application. Plug a headset mic into your PC, and you can make and take phone calls directly from VX Producer — no phone set needed. There's also a built-in audio editing application that lets you record, edit and play back conversations without the need for any third-party software. When you're done editing, click "Send to Studio" and it's ready to play on-air.

TELOS

INTEGRATED

PHONES ARE AN INTEGRAL PART OF YOUR SHOW. NOW, SO ARE YOUR PHONE CONTROLS.

Live calls or pre-recorded, interviews or audience participation, one thing's certain: phone segments are an integral part of today's fast-paced radio. But up to now, the phone system was separate from the on-air console; audio was shared, but little else. Wouldn't it be great if talent could take control of phones without ever having to divert their attention from the board? They can: IP-Audio networking technology provides the ideal way to integrate broadcast phones into the on-air console — the control center of every studio.

There are plenty of advantages to melding phones with consoles. Like ease of installation: IP-Audio consoles with built-in phone controllers don't need any additional wires or connections. Their control signalling, caller audio and backfeeds ride on the network connection that's already there. Information about line and caller status can be displayed right on the console as well.

Integration helps shows run smoother, too, since phone controls are right in front of them (instead of on some outboard phone control panel). With native phone controls on the console, talent can dial out, answer incoming calls, screen and drop calls in a more fluid, natural workflow than was ever before possible.

Flexibility is enhanced, allowing board operators to handle phone calls the way that's most comfortable for them. Bringing caller audio into the IP-Audio domain makes it routable like any other audio: for instance, talent's favorite Show Profile console snapshot might assign incoming phone lines to individual console faders for control with the familiar fader On and Off keys — or, they could have a hybrid assigned to a single fader and use the console's phone controls to quickly select between the incoming lines. With the Virtual Mixers built into Axia consoles, you could even choose to dynamically conference multiple lines and control their gain with a single fader. And since the console now communicates directly with the phone hybrid, mundane tasks such as mix-minus generation, starting recording devices, and playback of recorded off-air conversations can all be automated.

Naturally, Telos VX provides all the advantages listed here, integrating seamlessly with Axia Element 2.0 and iQ AoIP consoles, allowing talent to focus on doing what they do best — their show.

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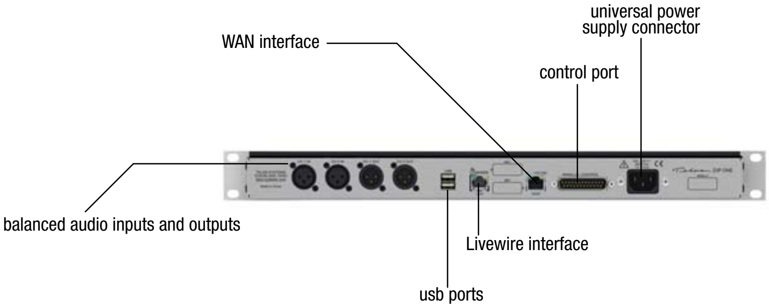
Z/IP ONE

TELOS



THE WORLD'S MOST ADVANCED 1 RU IP CODEC

Z/IP ONE

Z/IP ONE is the newest addition to the Zephyr family — an affordable, 1 RU codec designed to help you get the best possible quality from public IP networks and mobile phone data services, even from connections behind NATs and firewalls.

Agile Connection Technology (ACT), a Telos exclusive, is the foundation for the Z/IP's excellent performance on real-world networks. It delivers reliable audio despite varying network conditions, and without the need for user intervention. The Z/IP dynamically adapts to the network, minimizing the effects of packet loss and jitter.

When the network is well-behaved, you will benefit from the lowest possible delay and the highest possible fidelity. Should network conditions become challenging, the Z/IP ONE automatically responds by lowering the bitrate and increasing the buffer length, doing everything possible to ensure audio makes it to your studio reliably.

Z/IP ONE extracts excellent quality from even not-so-excellent IP connections, thanks to a new codec based on low delay AAC: Advanced Audio Coding-Enhanced Low Delay (AAC-ELD), which gives excellent fidelity at low bitrates with nearly inaudible loss concealment and very little delay.

And of course, Z/IP ONE speaks fluent Livewire; in addition to standard I/O, the Livewire connection lets it connect to any Axia IP-Audio network using just a CAT-5 cable.

- Z/IP ONE works with a variety of VoIP devices and can work with connections from compatible PBXs.
- Exclusive ACT (Agile Connection Technology) automatically senses network conditions and adapts codec performance to provide the best possible audio.
- Wide choice of standard high-performance codecs includes AAC-ELD, AAC-HE, MPEG Layer 2, G.711, G.722 and linear PCM.
- Slim 1 RU form factor fit is equally at home in a studio rack, remote kit or road case.

Z/IP ONE Front View

Available in four versatile models: 3 RU and 2 RU Studio Rackmount, the compact Z/IP Mixer, and the Z/IP ONE.

TELOS

THE WORLD'S MOST ADVANCED 1 RU IP CODEC

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Z/IP ONE

Z/IP ONE Back

Form Factor
- 1 RU Rackmount

User Interface
- Brilliant Front Panel OLED
- Remote Configuration and Control via Web Browser.

Connections
- Analog I/O (balanced XLR)
- Headphone Jack
- RS-232 Serial
- USB

SPECIFICATIONS

Audio

Input Levels - Menu Selectable	+4 dBu Pro -10 dBv Consumer
THD+N	< 0.01% @ +18 dBu, 1 kHz Sine
Freq Response	+/- 0.5 dB 20 Hz – 20 kHz
Headroom	18 dB
Dynamic Range	> 101 dB Unweighted > 103 dB “A” Weighted
Crosstalk	Crosstalk < -92 dB 10 Hz – 1 kHz < -79 dB @ 10 kHz
Output Clipping	1% @ 22 dBu Bridged Load (10.1 Vrms) 1% @ 22 dBu 600 ohm Load
Output Level Difference between bridged and 600 ohm Load	-0.67 dB
Calculated Output Impedance	45 Ohms < Zout < 50 Ohms differential
Calculated Input Impedance	10K Ohms < Zin < 14K Ohms differential
Analog to Digital Converter	24 bits
Digital to Analog Converter	24 bits
AES3 Digital Inputs and Outputs	Conform to AES3 Standards Sample Rates Supported (input): 32 kHz, 44.1 kHz, 48 kHz, 96 kHz Sample Rate Convert (output): 32, 44.1, 48 kHz; Slave to Input, Slave to Sync
Conformance and Compatibility	Conforms to N/ACIP (Open) Standards Fully supports Session Initiation Protocol 2.0 (SIP) Compatible with TCP, UDP, DNS, Zephyr Xstream, Uncompressed
Codecs	PCM and other Internet Protocols. SIP: G.711, G.712, MPEG Layer 2, MPEG AAC, Linear PCM MPEG AAC-Enhanced Low Delay (ELD) High Efficiency AAC

UNIVERSAL FEATURES

Network
Wired Ethernet
Wireless Wi-Fi, EVDO, UMTS

Signaling
Session Initiation Protocol 2.0 (SIP)
Direct or via Proxy Server.
Agile Connection Technology.

Codecs
AAC-ELD
AAC-HE
MPEG Layer 2
G.711
G.722
Linear PCM

Agile Connection Technology
Dial-By-Name, Group Speed Dial.
NAT/Firewall Traversal
IP Address Resolution
Media Relay
Adaptive Bitrate, Dynamic Buffering, Jitter Correction, Packet Loss Concealment.

Physical

	3 RU Rackmount	2 RU Rackmount	Z/IP Mixer	Z/IP ONE
Dimensions	19" x 5.25" x 15"	19" x 3.5" x 13.8"	18.4" x 4.5" x 15.63"	19" x 1.72" x 6.25
Height	3 RU	2 RU	2 RU	1 RU

TELOS

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Nx

Nx6 Package

TELOS

Nx

THE CLEANEST, MOST CONSISTENT CALL QUALITY EVER

SETTING NEW STANDARDS FOR STUDIO TELEPHONE SYSTEMS

Find a radio facility where caller audio quality is important, and chances are you'll also find a Telos Nx Talkshow System. Nx12 twelve-line and Nx6 six-line systems are the latest generation of Telos broadcast phone systems, built to deliver the cleanest, most consistent call quality possible from even the most challenging calls.

Nx systems combine multiple advanced telephone hybrids (each with their own AGC, noise gate, and caller override dynamics) with Telos' famous Digital Dynamic EQ, a sophisticated multi-band equalizer which analyzes and adjusts received audio spectral characteristics so that calls sound smooth and consistent despite today's wide variety of phone sets and connection types.

But there's more: Nx systems feature caller audio sweetening by Omnia, special echo cancellation to tame tricky VoIP and cellular calls, and anti-feedback routines to tackle the acoustic feedback that plagues open speaker applications.

Telos Nx6

Telos Nx6 and Nx12 systems can be ordered to work with your choice of POTS or ISDN (BRI) phone lines, and work with a variety of control surfaces, including the Telos Desktop Director, Call Controller and Console Director drop-in module. Of course, there's also an Ethernet connection for use with Telos Assistant Producer call screening software (and one-click connection to Axia IP-Audio networks).

With Nx Talkshow Systems, talent and producers both benefit from unique Telos features that help make shows run smoother, faster and more error-free, such as our exclusive Status Symbol visual call management icons that clearly show line and caller status. Both Nx6 and Nx12 can power as many as 4 control surfaces, or up to 8 surfaces using accessory power supplies.

Nx6 works with up to 6 telephone lines; Nx12 works with up to 12 lines. Both have four hybrids for extra flexibility in fast-paced talk environments. Nx6 and Nx12 feature a unique dual studio mode that allows a single system to power two studios simultaneously, analog and Livewire I/O, Program-On-Hold inputs; Nx12 can also be outfitted with an optional AES interface that allows direct access to all four hybrids individually.

Naturally, Nx Talkshow Systems connect directly to Axia IP-Audio networks using a single CAT-5 cable. One connection takes care of all audio I/O, on-hold inputs, hybrid control and GPIO. Drop-in modules available for Axia Element consoles let users easily take control of their Nx6 or Nx12 system right from the console. Choose the 2-Fader+Phone module with onboard hybrid controls with Status Symbol displays, or the 4-Phone module for a fader-per-hybrid European operating style.

Telos Nx12

Connections and Features

	Nx6	Nx12
Incoming Line Capacity	• (3) ISDN BRI (Sø or Uø), or • (6) Analog	• (6) ISDN BRI (Sø or Uø), or • (12) Analog
Hybrids	• (4) High Performance, Digital • Dedicate all (4) to one studio, or (2) each to (2) studios	• (4) High Performance, Digital • Dedicate all (4) to one studio, or (2) each to (2) studios
Audio Interfaces	• Analog • Livewire Audio-over-IP • AES (optional)*	• Analog • Livewire Audio-over-IP • AES (optional)*
Control	• Up to (8) Desktop Directors, Call Controllers, or Console Directors • Computer Controlled (ex. Telos' Assistant Producer Call Screening Application) via Ethernet	• Up to (8) Desktop Directors, Call Controllers, or Console Directors • Computer Controlled (ex. Telos' Assistant Producer Call Screening Application) via Ethernet

*Requires optional daughter board to repurpose XLR connections for AES.

Features

- Our most advanced hybrid algorithms.
- New symmetrical wide-range AGC and noise gate by Omnia.
- Studio adaptation and pitch shifter help prevent feedback in situations where open speakers are required.
- Adjustable caller override improves performance and allows you to individualize the degree to which the announcer ducks the caller audio.
- Digital Dynamic EQ and adjustable smart-level AGC by Omnia keeps audio spectrally consistent from call to call.
- Caller ID on both analog POTS and ISDN telephone lines, accessible over the Ethernet for call screening applications.
- Many control options to suit your individual requirements, both desktop and console-mounted.
- Status Symbols make life easier for producers and talent with their animated high-contrast icon display of line status.
- Input/output via analog XLR or Livewire Audio-over-IP. Optional AES3 I/O module.
- POTS, ISDN-S, or ISDN-U (specify when ordering) - Nx12 supports 50-50 mixed configurations.
- Two-studio mode for sharing one Nx and its telephone lines with two studios.
- When coupled with an Axia Element console, integrated control is possible using the Element's Call Control or Europhone Fader modules.
- Web server for configuration and remote monitoring.
- Flexible metering.
- Call Screening and remote control using Telos' Assistant Producer, or a wide variety of third party software packages.
- Function buttons and GPIO-style outputs for control of delay systems and recorders.

Nx

Specifications

Processing Functions

General
- Telos 3rd-generation Adaptive Digital Hybrids
- Telos Exclusive Feedback Reduction Functions, including Acoustic Echo Cancellation

Send (to caller) Processing
- Sample Rate Conversion
- High-pass Filter
- Studio side echo cancellation
- Frequency Shifter
- AGC/Limiter
- Pgm-on-hold AGC/Limiter

Receive (from caller) Processing
- High-pass 'Hum' Filter
- Phone line echo cancellation
- Noise Gate
- Smart AGC/Platform Leveler
- Telos' DDEQ (Digital Dynamic Equalization) 3-band Adaptive Spectral Processor
- Sample Rate Conversion

Analog Inputs

Send Analog Inputs	2x
Program-on-Hold Analog Inputs	2x
Connector	XLR Female, Pin 2 High (Active Balanced with RF Protection)
Input Level	Adjustable from -7 to +8 dBu (nominal)
Analog Clip Point	+21 dBu
Impedance	Bridging, > 10K Ohms
ADC Resolution	20 bits

Analog Outputs

Receive Analog Outputs	2x
Connector	XLR Male, Pin 3 High
Output Level	Adjustable from -7 to +8 dBu (nominal)
Impedance	<50 ohms
DAC Resolution	24 bits
Headroom Before Clipping	20 dB headroom from 4 dBU nominal levels

AES Digital Input/Output (Option)

Overview	Plug-in module converts the XLR inputs and outputs to AES3, providing 2x in on two XLR-Fs and 2x out on two XLR-Ms. The Nx12 provides 4x in on two XLR-Fs and 4x out on two XLR-Ms by using both channels of AES signal.
Rate Conversion	Sample Rate Converters on all inputs and outputs. Inputs can accept 32, 44.1, and 48 kHz rates. Clock for outputs may be sourced from the AES inputs or internally-generated, locked to the ISDN network.
Input Level	Adjustable from -27 to -12 dBfs
Output Level	Adjustable from -27 to -12 dBfs

Audio Performance

Frequency Response	±.5 dB, 50 to 20 kHz (swept sine procedure, measured from analog input to output with unit in loop-back mode).
THD+N/Input	< 0.06% typical (measured at +18 dBu @ 1 kHz, analog in to analog out in loop-back test mode).

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Nx

Nx12 Package

Switching Matrix and Conferencing

	Nx6	Nx12
Audio Routing and Switch	All Digital	All Digital
Telephone Lines	6	12
Hybrids	4	4
Studio Inputs	4	4
Studio Outputs	4	4
Program-on-hold	2	2

Control Ports

Ethernet 100Base-T

- Web server for configuration and software update
- Telnet for command line control and diagnostics
- Assistant Producer server allows up to 8 instances of Telos Assistant Producer (v3.5 or later) to connect simultaneously

General purpose Input/Output

The Nx6 has two 9 pin D-sub, the Nx12 has one 15 pin, with status outputs and control inputs.

ISDN Telephone Connectivity

Protocol Compatibility

- National ISDN 1 and 2
- DMS-100 Custom Function
- AT&T 5ESS Custom Point-to-Point
- Euro-ISDN conforming to the Net 3/ETS300 Protocol

Interface

- USA & Canada: Integrated NT1 for direct connection to ISDN line via the two-wire U-interface (6-position/2-pin RJ-11 connector) 2B1Q Line Encoding
- Worldwide: 4-wire S-interface (8-position/8-pin RJ-45 Connector)

Telephone Coding Modes

- µLaw (ISDN Proto set to Natl I-1, AT&T Custom, Q.931mu or DMS Custom)
- A-Law (ISDN Proto set to ETS-300)

Analog Telephone Line Connectivity

- Universal interface for worldwide application
- Programmable loop current
- Programmable ring and disconnect signaling (loop drop or tone)
- Programmable Flash time
- Caller ID decoding using Bellcore 212 modem standard

Desktop Director Ports

- Eight directors using external RJ-45 splitters and power. (Extended Desktop Directors count as 2 for power purposes)

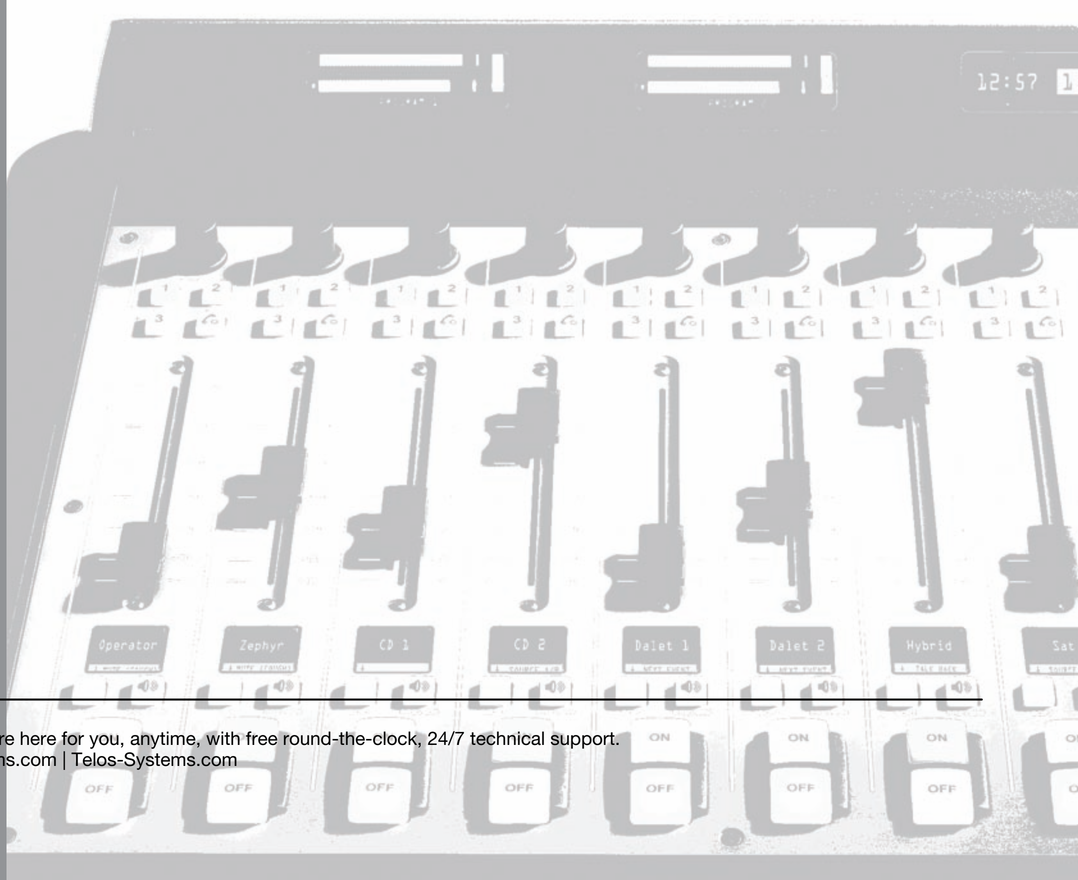
iQ6

iQ6: MULTI-LINE PHONES MADE SMARTER

Six lines, one tiny connection. iQ6 is the Telco gateway for Axia's new iQ console. iQ6 saves you money and time, because it plugs right into your Livewire network, eliminating the cost of discrete I/O, cabling and soldered connectors, and drastically reducing installation time.

iQ6 carries six lines worth of audio, hybrid control and backfeed on one skinny CAT-5 cable. You control it directly from Axia iQ consoles equipped with the Telco expansion frame, or by using the new Telos VX Director or VX Producer software to take and screen calls directly via your producer's PC.

iQ6 works with POTS or ISDN phone lines, and comes equipped with two 3rd-generation Telos hybrids with Digital Dynamic EQ; the same advanced hybrids found in our Nx phone systems. iQ6 also comes with new AEC - Advanced Echo Cancellation - from Fraunhofer Labs, revolutionary technology that eliminates open-mic feedback.



Hx digital hybrids

Hx1

Hx2

Hx1 AND Hx2
DIGITAL HYBRID TELEPHONE INTERFACE

In the mid-1980s, Telos pioneered the very first digital adaptive telephone hybrid. Since then, our POTS phone hybrids have earned a worldwide reputation to deliver clean, clear caller audio from even the most difficult calls. The new Telos Hx1 and Hx2 represent the highest state-of-the-art in hybrid performance, with Telos processing technologies that take the POTS hybrid to a new level of consistently superior performance, regardless of telephone line characteristics.

Inside the single-hybrid Hx1 and dual-hybrid Hx2, you'll find our most advanced hybrid technology with new features that sweeten and control caller audio better than ever before. Hx hybrids come standard with features you won't find in other POTS hybrids, like Auto-Answer, caller disconnect detection, sophisticated new audio-leveling and anti-feedback routines for enhanced open-speaker applications, call screening and line-hold features, and front-panel send and receive audio metering — plus much, much more. (Just take a look at the specs below and you'll see what we mean.)

Incoming Line Capacity	• (1) POTS Analog for Hx1 • (2) POTS Analog for Hx2
Hybrids	• (1) High Performance, Digital for Hx1 • (2) High Performance, Digital for Hx2
Audio Interfaces	• Analog • AES (optional for Hx2)
Ancillary	• Single 9-pin D-Sub connector to remotely turn each hybrid ON/OFF and to return Ringing and On-Air status for each hybrid

Hx digital hybrids

FEATURES

- Our best, most advanced hybrid algorithms.
- New symmetrical wide-range AGC and noise gate by Omnia, with adjustable gain settings.
- Studio adaption and pitch shifter help prevent feedback in situations where open speakers are required.
- Adjustable caller override improves performance and allows you to individualize the degree to which the announcer ducks the caller audio.
- Digital Dynamic EQ and adjustable smart leveler keeps audio spectrally consistent from call to call.
- New EQ High and EQ Low display meters for each hybrid.
- Separate Send level and Receive level meters for each hybrid.
- Status Symbols make life easier for producers and talent with their animated high-contrast icon display of line status.
- Place caller on-hold via front panel button.
- Auto-Answer with selectable ring count.
- Worldwide disconnect signal detection. (loop drop, dial tone, or re-order tone)
- Input/output via analog XLR. (Optional AES3 I/O module for Hx2)
- Input switchable between MIC or LINE levels.

Hx1 back view

Hx2 back view

Processing Functions

General	• Telos 3rd-generation Adaptive Digital Hybrids • Telos Exclusive Feedback Reduction Functions, including Acoustic Echo Cancellation
Send (to caller) Processing	• Sample Rate Conversion • High-pass Filter • Frequency Shifter • AGC/Limiter
Receive (from caller) Processing	• High-pass 'Hum' Filter • Smart AGC/Platform Leveler • Noise Gate • Caller Ducking • Telos' DDEQ (Digital Dynamic Equalization) 3-band Adaptive Spectral Processor • Sample Rate Conversion

TELOS

Hx digital hybrids

Analog Inputs

Send Analog Inputs	• 1 for Hx1 • 2 for Hx2
Connector	XLR Female, Pin 2 High (Active Balanced with RF Protection)
Input Range	Select between MIC and LINE levels
Input Level	Adjustable from -10 to +4 dBu (nominal)
Impedance	Bridging > 50K Ohms
Analog Clip Point	+21 dBu
Analog-to-Digital Converter Res	24 bits

Analog Outputs

Receive Analog Outputs	• 1 for Hx1 • 2 for Hx2
Connector	XLR Male, Pin 3 High
Output Level	Nominal at +4 dBu
Impedance	< 50 Ohms
Digital-to-Analog Converter Res	24 bits
Headroom Before Clipping	20 dB headroom from 4 dBu nominal levels

AES Digital Input/Output (Hx2 Option)

Overview	Plug-in module converts the XLR inputs and outputs to AES3 (one input or output on Left channel of AES stream)
Rate Conversion	Sample Rate Converters on all inputs and outputs. Inputs can accept 32, 44.1, and 48 kHz rates. Clock for outputs may be sourced from the AES inputs or internally-generated 48 KHz
Input Level	Nominal at -20 dBFs
Output Level	Nominal at -20 dBFs

Audio Performance

Frequency Response	200 to 3400 Hz, +/- 1 dB
THD+N/Input	< 0.5% THD+N using 1 kHz sinewave
Signal to Noise	> 90 dB

Control Ports

General purpose Input/Output	Single 9 pin D-Sub connector with 2 status outputs (Ringing and ON-AIR) and 2 control inputs (ON and OFF) per hybrid

Analog Telephone Line Connectivity

- Universal POTS interface for worldwide application
- Programmable loop current
- Auto-Answer with selectable ring count
- Worldwide disconnect signal detection (loop drop, dial tone, or re-order tone)

TELOS

MORE STUFF GALORE

Telos's line of telephony products doesn't stop here. Below are some of our other popular products.

Z/IP Rackmount & Mixer

Zephyr/IP offers high quality, reliable, audio-over-Internet for remote broadcasts and connecting remote studios. Featuring Livewire, AES/EBU and stereo analog audio I/O with 24 bit AD/DA converters. The Z/IP transmits and receives high quality, stereo or mono audio using any of several coding algorithms, including AAC-ELD at bitrates from 16 to 256 kbps. Reliability over non-QoS IP connections is enhanced using Telos' exclusive Agile Connection Technology (ACT). The free, private Z/IP server allows easy connections to other Z/IP units through firewalls using user-friendly names. The live status of other Z/IPs is displayed in a speed dial list. Connect via wired Ethernet or wireless with approved USB Wi-Fi and 3G cellular devices. The Z/IP Mixer combines the versatility of the Zephyr/IP with the utility of a digital four-channel stereo mixer, all in a rugged, road-ready portable chassis.

Zephyr iPort

Zephyr iPort enables broadcasters to transport multiple channels of stereo audio across IP networks with guaranteed QoS, such as T1 and T3 connections, MPLS networks, et cetera. Can be configured for use either as a codec to transport eight stereo bi-directional MPEG streams, or as an MPEG encoder for up to 16 stereo output streams. Connects to your Axia Network using a single CAT-6 cable for all I/O — or pair with an Axia AES or Analog Audio Node for use as a standalone multiple-stream codec. Requires one unused 100Base-T Ethernet port for connection to an Axia network. Autosensing power supply, 90 VAC to 240 VAC, 50 Hz to 60 Hz. 100 Watts. Rackmount, 2 RU.

Zephyr Xstream

World's leading ISDN codec, compatible with the widest variety of third party codecs. Includes MPEG-AAC and AAC-LD, Layer 2, Layer 3, G.722 coding, AES/EBU, Livewire IP, ISDN TA with integral NT1 supports worldwide compatibility without changing software. Features full duplex stereo operation of up to 20 kHz audio on a single ISDN line; broadcast quality mono audio at 15 kHz or 20 kHz is possible on a single ISDN "B" channel or other 56/64 kbps channel. Remote Control over RS-232 or Ethernet. Single headphone output. The Xstream MXP houses the Xstream in a rugged shock resistant case that helps it stand up to the rigors of the road. Comes equipped with a flip up stand for best viewing angle. Four-channel stereo mixer includes selectable digital processing by Omnia.

Zephyr Xport

Portable unit allows use of an ordinary analog telephone line in the field to connect with the Zephyr Xstream ISDN codec at the studio. Xport features the highest fidelity low-bitrate coding available, AAC plus (MPEG AAC + Spectral Band Replication enhancement). Features include: Mixer with mic and line inputs, internal power supply and selectable DSP processing by Omnia. Can be upgraded to POTS & ISDN using optional ISDN interface, supporting AACLD and G.722. The POTS+ISDN version includes ISDN Interface with Internal NT-1. Includes RJ-45 "S" for use worldwide and RJ-11 "U" connectors for use in the USA and Canada.

Desktop Director

Telos Desktop Director gives talent intuitive control of Nx12, Nx6, 2x12 or Series 2101 systems' conferencing, call screening and other talk show functions. Standard Director controls 12 lines and two hybrids; Extended Director works with Nx12 or Series 2101 systems to control up to 24 lines and two hybrids, or 12 lines and four hybrids. Includes built-in acoustically isolated speakerphone; headset port accommodates your favorite third-party screener's set.

Call Controller

Cost effective Call Controller works with Telos Nx12, Nx6 or TWOx12 Talkshow Systems to provide line status and switching for up to 12 lines and two hybrids. Connect the Call Controller to a POTS phone of your choice for screener or studio telephone operation.

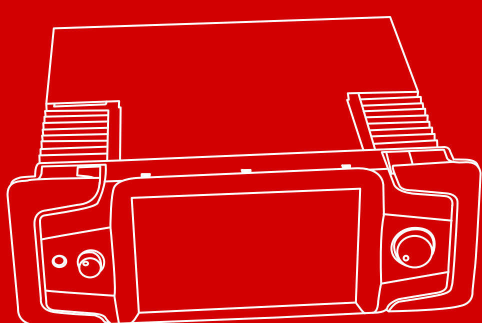
Assistant Producer

Assistant Producer for Nx12, Nx6, TWOx12 and Series 2101 allows Producers and Talent to manage callers while maintaining constant communication with each other in an easy-to-use Windows^{TM} environment.

TELOS

what's new! /omnia / 11

what's new! /omnia / 11



THE BIRTH OF OMNIA.11:
MORE REVEALING.

YOU CAN'T USE JUST ANY PROCESSOR TO DO THE JOB OF AN OMNIA.

A question was posed to Frank Foti, a few years ago, by an editor of the trade press. The proverbial, "where does Omnia go from here?" Frank's reply was simple, "Our next offering must restore quality back into the on-air signal, while maintaining competitive texture." Easier said than done. Yet once again, he and his team have delivered!

Omnia.11 is not just the next step forward from our prior offerings. We took our platform, stripped it down beyond the bare walls and built a fortress. Everything from the firmware platform, the GUI, to every algorithm was re-thought, and in most cases re-engineered or re-designed. The signature Omnia sound was kept intact, while the level of perceivable quality has noticeably improved.

The firmware in Omnia.11 takes advantage of software resources never before available for this calibre of product. The results are dynamics algorithms that were once a pipe dream of the processing enthusiast. Instead of making all the clichéd claims about how much better it sounds than such-n-such box (we'll leave those for others to make) you will hear how Omnia.11 is more revealing… period.

OMNIA.11 TECH OVERVIEW

Chameleon Processing Technology
AGC's, Compressors, and Limiters analyze music in real time and adjust internal parameters for optimum performance across a broad range of material. Listeners will hear the music, not the processor.

A major part of this technology, the new Density Detector, enables Omnia.11 to properly handle hypercompressed content. The AGC system cannot be fooled due to heavy density, or older material that contains high peak-to-average levels. One could almost say, there isn't a sweet-spot, as the density-detector keeps Omnia.11 operating on-target, at all times.

Ultra-LoIMD Multiband Limiter System
Limiters have traditionally carried "that bizzy limiter sound". You know, when drums sound like tin cans and detail gets smashed to smithereens. Our new LoIMD technology coupled with smart gain reduction algorithms resets the bar. The limiters now sound amazingly transparent.

All AGC and limiting algorithms employ an auto acceleration/deceleration mechanism, which tunes out perceptible intermodulation distortion. The attack/release functions adjust themselves based upon content density. This breakthrough method literally analyzes the audio content in both the amplitude and frequency domain, then adapts the timing networks – on the fly – to transparently control the signal, without the control being heard. The result is revealed in added detail, clarity, and quality, yet maintaining the desired competitive loudness level.

Needless to say, the improved performance of the AGC and limiter functions generate live voice quality that is truly second to none. We'll say no more.

Bass Management
Boom-Boom, out go the lights! The bass enhancement algorithm is audio processing's testosterone. We took our Phat Bass effect and sent it to a health club, resulting in low end physique with no body fat. Full punch, power, and girth, with no nasty side-effects whatsoever. New material, old material, Rock, Country, Soft A/C, Urban, Fine Art, Hip-Hop, Pop, Talk Format... you name it, this is HGH (human growth hormone) for processing!

OMNIA

All kidding aside, our new bass-management method is a mixture of innovation, as well as a re-arrangement of the system topology. Achieving great sounding bass requires the most effort, partly due to the fact that the bass spectrum has the most number of harmonics, and all of these must be kept properly accounted for in the time domain. Also, any additional spectra created (enhancement) must have its harmonic content managed, or the bass region begins to sound distorted, unnatural, or like someone is suffering from an intestinal disorder! This process requires much more than just fancy EQ, bass clipping/filtering, or any of the prior attempts at bass enhancement. Even the location where the function is inserted matters, as well as how it maintains its frequency range along with the rest of the system. An entire dissertation could be done on the bass enhancement/management system alone. Our classic dynamically flat & time aligned crossover system has been further refined to produce smooth, rich, and full tonality. The AGC and limiter sections cannot be fooled into false gain control due to spectral density (or lack thereof) from the crossover network.

New Ultra LoIMD Distortion Controlled Clipper System

Audio processing for conventional broadcast (FM and AM) has reached extreme levels. Various methods are available today capable of creating LOUD competitive signals, but at the expense of perceptible quality. What causes this, and what can be done - again - to raise the bar? Through critical listening, extensive research, and evaluation of processing methods, it has been determined the single most annoying quotient is due to intermodultion distortion (IMD) induced by aggressive functions within the processing system. The algorithms are pushed to the limits, and beyond. One of the most crucial, agressively used algorithms in the FM processor is the pre-emphasized final limiter/clipper. We have developed the new Ultra LoIMD distortion controlled clipper system specifically to reduce IMD in this critical stage of the processing. See the next page for a full explanation of the new Ultra LoIMD clipper system.

For those who feel the need to use it, there's also a composite clipper embedded in the stereo generator, but thus far, all of our testing has been done without any composite clipping. Pilot protection is on the order of magnitude close to 90 dB, considerably more protection provided than the best FM receiver can recognize! Integrated laboratory-grade stereo generator with dual MPX outputs, 19 kHz reference output for external RDS/RBDS systems and pilot protection that provides >80 dB pilot protection – with or without composite clipping. MPX spectral low-pass filter to protect RDS/RBDS and SCA signals if composite clipping is employed.

Adjustability

Multiple ways to adjust the system to achieve the exact sound you're looking for. An installation wizard will guide anyone through a simple step-by-step setup to on-air operation. Using the answers to a series of simple questions, Omnia.11 adapts itself, based upon the answers, to craft a preset which delivers the desired end result quickly for an advanced out of the box experience.

Unprecedented Access

A front panel touch screen GUI, on a 10.5" diagonal screen, provides ease of use and enhanced metering and diagnostics. Remote access is via any web browser, as well as a local onboard WI-FI connection. Laptops to iPads have access!

Livewire, AES/EBU digital and analog I/O is standard. Headphone soft "patch points" are available for listening through the processing chain.

Fanless cooling design built into a rugged 4 RU chassis.

ADDITIONAL FUN STUFF

Diversity-Delay, with ramp-in and ramp-out ability, for smooth transition and alignment of conventional analog to HD-Radio signals. (Max delay is 20 seconds)

ITU-BS-412 MPX limiter.
Dual processing paths for FM and coded audio transmissions.

FEATURE PACKED FUTURE

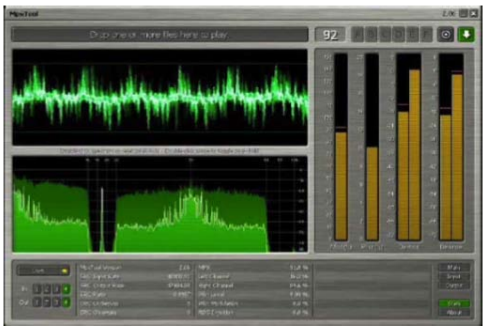
Optional RDS/RBDS generator.

Optional HD Radio time alignment method.

Optional Third-Party plug-ins for enhanced processing library/toolbox.

Optional Arbitron PPM encoder and confidence monitor. We have embedded the PPM encoder where it belongs, in the audio processor. Just as we initiated the diversity-delay function for HD Radio, Omnia innovation leads the way - again - in the broadcast transmission world. The PPM confidence monitor does more than indicate when the PPM signal is not present. It can email or text the end-user to alert them of a PPM situation.

FEATURE PACKED FUTURE (cont)



MPX Tool… a very cool tool developed by Leif Claesson. This embedded application displays the FM multiplex (MPX) signal, peak modulation, pilot level, as well as a preview of the deviated FM carrier. A built-in complete transmission signal analysis system right at your fingertips!

Also currently in development are additional tools to measure quantifiable processing elements like: fatigue, distortion, quality, clipper depth and power in each processing band, along with utilitarian functions to log preset history, and detailed system diagnostics.

ULTRA LoIMD CLIPPER SYSTEM EXPLAINED:

Audio processing for conventional broadcast (FM and AM) has reached extreme levels. Various methods are available today capable of creating LOUD competitive signals, but at the expense of perceptible quality. What causes this, and what can be done - again - to raise the bar?

ON-AIR ANABOLIC STEROIDS

Audio processing is no different today than athletes who take performance enhancing drugs to get an extra edge. A scan across the dial in most markets, around the world, yield radio stations transmitting DC modulation as their means to be the loudest/proudest on-the-air. Some even subscribe to the notion of "more loudness than allowed by law!"

Not to be outdone, the music industry has adopted the same radio mindset producing L-O-U-D CDs that are so heavily processed, they sound as if they're on-the-air, before being on-the-air. To say we live in a dynamically processed world is an understatement!

Being loud is not the problem. "If it's too loud, you're too old" does apply. The problem is the unfriendly annoying artifacts generated by current processing practices used by broadcasters and the music industry. The combination of hyper-compressed content and "I gotta be louder than the other guy" on-air processing, results in audio lacking definition, and quality, while containing perceiving annoyances. It would be easy to say, just back down the processing and all will be OK, except reality, and the psychology of broadcasters suggest differently. We will always have loud radio stations, as long as programming philosophy remains, as it has, ever since Mike Dorrough fathered multiband processing.

The challenge now is how do we put quality back into on-air audio; yet retain the competitive loudness level broadcasters demand?

FINDING THE ACHILLES HEEL

The path to find the next frontier in processing began with a simple, but somewhat frightening, exercise: careful and extremely critical listening of the status quo. As a developer it requires leaving your own ego at the door and being prepared to face some harsh realities. Suffice it to say, we were about to embark upon a humbling experience.

Having been around the block a few times with respect to loud competitive processing, our own system was setup to simulate any one of a number of well-known major market radio stations. Hot Rockin' Flame Throwin' applies! Benchmarking was done using stock processing and cranked up presets, along with the use of processors from other manufacturers. The choice of programming content was made up of recent recordings as well as many time-tested segments used over the past twenty-plus years for processing development.

THE PAIN

Modern music mastering practices generate content that is noticeably rich in deep bass, presence, and treble. When processed aggressively, especially for FM-Stereo, the resulting audio appears synthetic in tonality and quality. Treble frequencies appear overly bright, and sometimes harsh, even with additional application of dynamic high frequency processing. Bass sounds tight and defined, but depending upon the processor's spectral limiting system, it can also sound distorted.

Listening to current music, with aggressive processing, produces a distinct annoyance: the appearance of a sizzling or frying sound to midrange, presence, and treble spectra. This was noticeable on all the processors used for evaluation. Reducing the final limiting, or clipping, helped ease the pain. This indicated that the problem is harmonic, related to the clipping process. Significant reduction of clipping removed the annoyance, but the loss of loudness was on the order of 6 dB or more. Not suitable for the needs of competitive audio.

Was this the age-old issue of too much bass forcing the rest of the spectrum into the limiter? This is known as bass induced intermod. At first, it would seem so, but the test segments did not have any bass content, and the frying was still present. Was it in the original source, and the processor was magnifying it via multiband dynamics control? Careful evaluation of the source audio revealed the answer to be no. Seems a new gremlin hath arrived!

By example, this is easily heard in the song "Because Of You" by Kelly Clarkson. The opening of the selection is a piano solo, and the vocalist begins to hum along with the piano, a few seconds later. Present day audio processors, set up aggressively, cause the humming in the vocal to sound as if bacon is being fried! Prior statement not intended for humor, by the way. This was a high rotation song on CHR (Contemporary Hit Radio) formatted radio stations. Since most of those require aggressive processing, this test case replicates the real world. This example is just one of many which illustrate the challenges in current processing technique.

THE INVESTIGATION

Since the aforementioned bacon frying annoyance was affected by the action of the clipping function, a probe into that algorithm was in order. Most final limiting/clipping systems in modern audio processors employ some form of proprietary means to control perceived distortion. The methods for these vary. While open for subjective discussion, the end result is still the same: absolute peak control is performed and a minimal level of harmonic distortion is acceptable within a specified operating range. Basically, some form of masking method is used to hide the most annoying clipping side effects from the ear. Although, it appears now, we've pushed these methods to the point where modern recordings generate distortion annoyances when aggressive processing is used.

For the processing novice, a clipper – by design – will generate harmonics of the fundamental audio frequency. Using a sinewave for an example, if the upper and lower peaks of the waveform are chopped off (clipped), harmonics are born out of the clippings and show up within the spectrum space as harmonic multiples of the original frequency. An example frequency of 1 kHz, with 3 dB of clipping generates odd-order harmonics at 3 kHz, 5 kHz, 7 kHz, etc, out to infinity. Figures 1 – 2 illustrate examples of this.

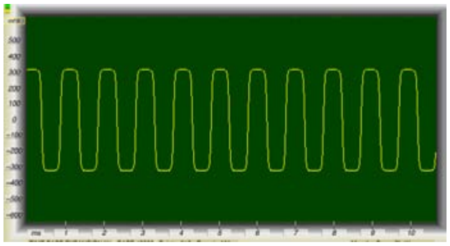


Figure-1, 1 kHz Square Wave

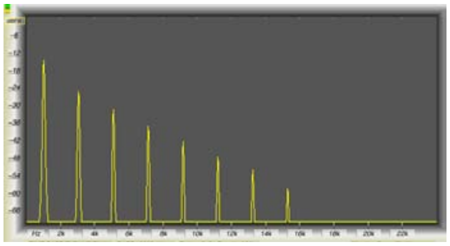


Figure-2, 1 kHz Square Wave Spectra

Note: Broadcast audio processors band-limit frequency spectra within a specified range, for their respective transmission paths. FM-Stereo bandwidth is 15 kHz, and AM is between 4.5 kHz and 10 kHz, depending on location. As such, clipping harmonics are limited via non-overshooting filtering methods in order to properly maintain operating legal bandwidth.

The challenging problems stated here are not based upon clipping functions of singular frequencies. Modern clipping methods, with distortion management, reduce clipping side effects over a preset range and only up to a specified level. It appears modern recording techniques either overload the present distortion mechanisms, or they cannot process this content aggressively without generating this frying/sizzling-like distortion. Since this problem exhibits itself with full range linear recordings, data reduced content (mp3 audio files) is even more distorted. This question becomes, where's the rub?

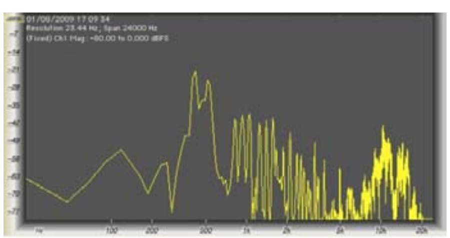
WHAT HAPPENS WHEN ADDITIONAL SPECTRA IS ADDED?

When additional audio is added to a fundamental, lots of fun activity occurs! Sum and difference frequencies are created along with another component known as intermodulation, or IMD for short. Simply stated, this is where one signal will ride alongside, on top of, or modulate another. Sometimes this is done for specific effect. Music synthesizers use various intermodulating functions to create desired sounds.

In an audio processor, the dynamic action of compressors and limiters are examples of modulators, as they generate a level controlling signal to change the gain of the audio. The level controlling signal and audio is routed to a multiplier function, and the audio is multiplied by the controlling signal. Through this action, the level is dynamically adjusted. This is an example of intermodulation, as the audio is modulated by the control function. When the control signal starts to operate too fast, it generates a controlling rate with an additional frequency of its own. This operating frequency will possess additional harmonics and those get factored (multiplied) into the audio during the multiplication stage. The resultant contains the level adjusted audio along with harmonics from the controlling signal that were intermodulated into the final product. This is what happens when the control signal operates in an overly aggressive manner: the sonic quality becomes fuzzy, dull, and lifeless. We refer to this as dynamic intermodulation distortion.

With the above example in mind, let's consider what happens within a clipper, when multiple audio signals are present and clipping is applied. A clipper, in reality, is a zero-attack/zero-release time limiter operating with a ratio of infinity-to-one. When multiple frequencies are present and clipping is active, the lower fundamental frequency will push the higher fundamental frequency into, and out of, the clipper at the rate of the lower frequency. This is known as clipper induced IMD. An easy example of this would be music with deep defined bass and a solo guitar or vocal. When clipping is active, the guitar or vocal will warble at the rate of the bass frequency due to the action of the bass signal pushing the guitar/vocal signal in and out of the clipper. Some audio processors employ bass processing techniques to reduce - and in some cases - remove this annoyance. On account of this, IMD components are amplified in level and spectra. Even modern distortion cancelling clippers (or whatever other marketing name is given to them) generate IMD.

Up until now, it's been an accepted notion that clipper induced IMD was a by-product of deep bass and enhanced midrange/presence/treble content. When studying the example of the Kelly Clarkson track, it became evident the problem was related to clipper induced IMD, except the example does not possess any bass spectrum of any significance.



Notice in Figure-3, a segment taken from the Kelly Clarkson track, the dominance of signal centered at 500 Hz, and the range between 10 kHz – 15 kHz. Wonder what happens if some IMD tests were run on present clipping systems?

UNDER THE MICROSCOPE

Performing an IMD test on a clipping system is quite easy. Two audio frequencies are mixed together, then passed through the system under test and the output is observed on a scope and spectrum analyzer. In this instance, the clipping systems all employed the required 15 kHz low pass filtering and zero-overshoot control mechanisms found in broadcast processors.

For the test, 100 Hz was inserted at a level, which generated 3 dB of clipping. A high frequency component was mixed in at the same level and 75 μs pre-emphasis was applied. The tests were run over the range of 5 kHz up through 15 kHz, while 100 Hz was used as a constant low frequency source. Figures 4 – 8 are the results of the tests.

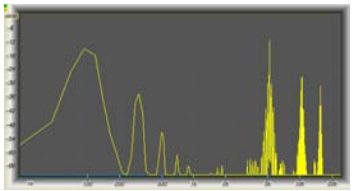


Figure-4, Clipper Induced IMD: 100 Hz & 5 kHz

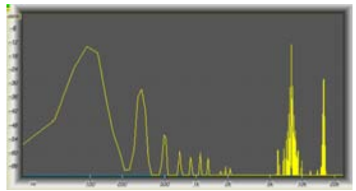


Figure-5, Clipper Induced IMD: 100 Hz & 7.5 kHz

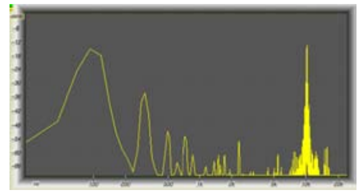


Figure-6, Clipper Induced IMD: 100 Hz & 10 kHz

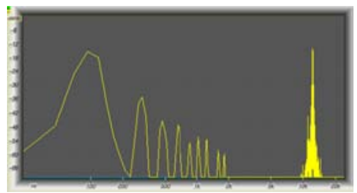


Figure-7, Clipper Induced IMD: 100 Hz & 12.5 kHz

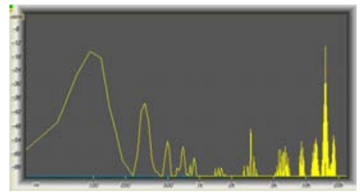


Figure-8, Clipper Induced IMD: 100 Hz & 15 kHz

Notice as the upper frequency is increased, there is significant difference spectra that falls between the two fundamentals. This is extremely severe at 10 kHz, 12 kHz, and 15 kHz. If you recall the music example, this is very close to the spectral illustration in the Kelly Clarkson track. Houston, we've got a problem! It is clipper induced IMD!

CLIPPER SYSTEMS, DISTORTION CANCELLING, AND TOO MANY BANDS!

As stated, all present day clipping systems employ methods to control distortion. Of interest is that each of these use a static method to mask harmonic distortion when clipping is active. As the Kelly Clarkson example clearly illustrates, harmonic distortion is not the concern as it once was. Intermodulation, due to added presence and high frequency spectra, has overtaken the problem that once was dominated by harmonic distortion. Suffice it to say, all clipping methods must employ some form of harmonic distortion control, or they will not operate sufficiently enough to generate competitive sounding on-air audio. Modern content now requires additional processing means to reduce induced IMD.

Suppressing IMD is significantly more difficult, as the constantly different frequency components are a non-stop moving target. Whereas suppressing harmonic distortion can easily be predicted and controlled through a static filtering system.

Proof of this is demonstrated with an evaluation of present day distortion cancelling systems. All of them employ static filtering to mask distortion components. They vary in range from broadband to 5-6 band, or more. All of these fail with aggressive processing. The broadband method suppresses harmonics and some IMD at specific frequencies. The multiband methods are designed to insert gentle low pass filters after multiband clippers in each audio band. This works over a narrow range, but falls apart with aggressive levels of clipping. Multiband clipper/filtering is done in parallel architecture and each singular band clipper is not able to understand what the others are doing. Therefore, the resulting filtered harmonics of each band interact in unpredictable ways - some of which exaggerate IMD. Adding more bands or steeper filters does not improve or fix the problem!

THE ANSWER IS...

… not in the number of bands. Anyone who thinks adding more bands of clipping and filtering to the system is wasting DSP cycles, or computer MIPS, along with wasting your time with a lot of marketing rhetoric. The answer lies in understanding the range of frequencies that generate both harmonic and intermodulation distortion, then applying various masking means to suppress both simultaneously as they are generated. Much easier said than done! It's a combination of breaking down the audio spectrum by octaves and interaction with the Gibbs Phenomenon. Almost explained too much already. Suffice it to say, the prior statement - along with SENSUS technology - enables a clipping system that suppresses BOTH harmonic and IMD distortion components when aggressive processing levels are required. Additionally, and more importantly, this new clipping method does not employ the use of dynamic compressors or limiters to control depth of clipping in order to minimize clipping induced IMD. There have been, and remain, a few proponents who utilize this method to reduce generated IMD, but it is at the expense of added dynamic intermod which manifests itself as audio pumping and hole punching.

PROOF

Running the same IMD tests, as mentioned earlier, now offer the following results. Compare figures 9 - 13 to those of figures 4 - 8 of both the old and new methods.

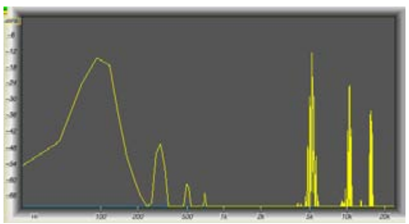


Figure-9, Clipper Induced IMD: 100 Hz & 5 kHz

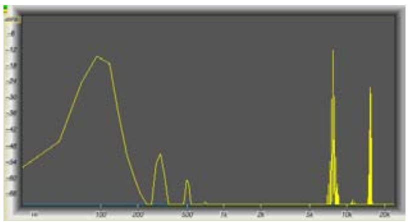


Figure-10, Clipper Induced IMD: 100 Hz & 7.5 kHz

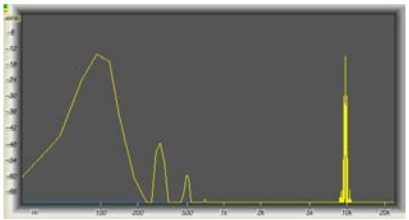


Figure-11, Clipper Induced IMD: 100 Hz & 10 kHz

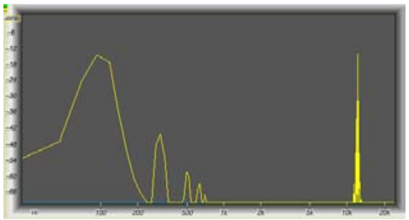


Figure-12, Clipper Induced IMD: 100 Hz & 12.5 kHz

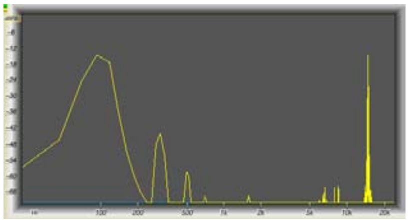


Figure-13, Clipper Induced IMD: 100 Hz & 15 kHz

It is easy to see. For the exact same amount of clipping employed, midrange, presence, and treble IMD is gone! With the new method, Kelly Clarkson's test segment does not possess any of the bacon frying sizzle annoyance as heard prior with all other clipping systems. As a matter of subjective observance, all audio auditioned through this new method sounds cleaner for the same given level of loudness. It does not matter if the content source contains deep-rich bass or not: the audio signal is subjectively cleaner for the same level of loudness.

WHAT GIVES?

Just the same as with the recipe for Coca-Cola®, this new IMD and THD masking method is locked away in the same vault with the secret ingredients for the famous soft drink! What can be said however, is the answer lies not in the number of clipping bands and filters, but how all distortion products are dealt with interactively on an instantaneous basis. Multiband clipping does not take into consideration any interactivity of outlaying spectra. That's where the method eventually fails. The proof is in the audio performance with critical content.

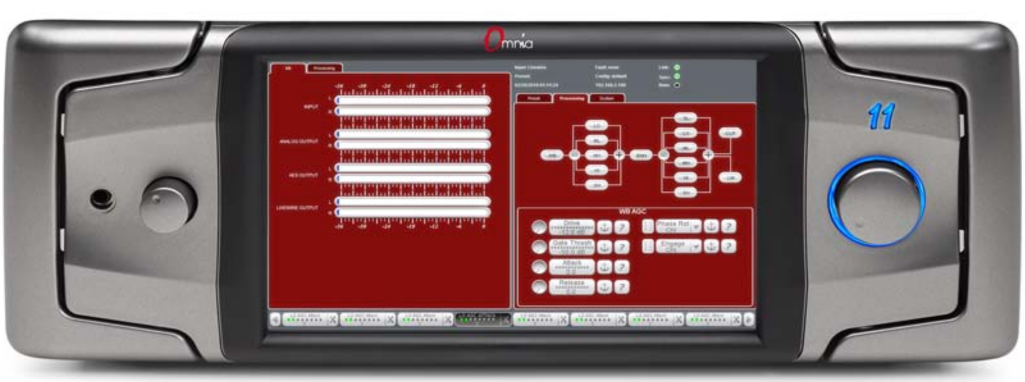
ACKNOWLEDGEMENTS

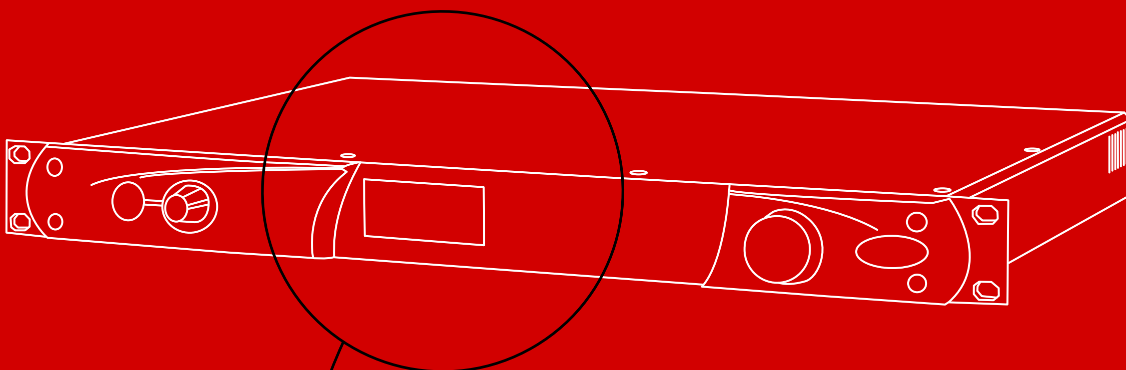
This is dedicated to the loving memories of Jim Somich and Mathew Connor. Two passionate, dedicated, driven, and innovative engineers who understood great sounding radio, how to get there, and the advantages of "dial-dominance!" You both are truly missed.

Additionally, Rob Dye, Leif Claesson, Tim Carroll, Mike Dorrough, Cornelius Gould, Mark Manolio, and Steve Church have contributed significantly to the success and growth of our Omnia product family.

Modestly speaking, we haven't been sitting around putting a fresh coat of paint on an age old – and vastly tired – design. Omnia.11 is designed for today and the future. Consider that we stretched the Omnia.fm firmware platform close to fifteen years, as that design grew into Omnia-3, Omnia-5 and Omnia-6. Our Omnia ONE platform re-design lent itself to become the tech basis of Omnia.11. An investment in Omnia is more than an investment in quality, competitive on-air audio today. Our track record proves your investment is safe for the foreseeable future both sonically and financially.

You can't send just any processor to do the job of an Omnia!





what's new! /omnia /

ONE

what's new! /omnia /

ONE

THE ONE PROCESSOR THAT DOES IT ALL NOW DOES MORE

The Omnia ONE meets the challenges of AM, FM, HD Radio™, DAB, DRM, multicasting, podcasting, netcasting, satcasting, and just about any application where signal processing is needed. The Omnia ONE quickly adapts to your needs by simple software downloads. It can be used in the studio, for transmission, and even in networked applications, thanks to its Livewire interface. It even has dual software banks allowing you too easily switch between styles.

The Omnia ONE packs an impressive list of standard features into its 1 RU frame:

- Wideband AGC followed by Four-Band AGC and Four-Band Peak Limiter sections. +
- Browser-based remote control and configuration. +
- Analog XLR balanced inputs and outputs. +
- Digital AES/EBU input, output and external Sync input. +
- Automatic input fail-over on loss of audio. +
- Livewire/Ethernet RJ45 jack. +
- Universal Power Input. +
- Built-in stereo headphone jack with front panel level control. +
- Single Jog-Wheel user interface with LED level metering and LCD screen. +

> **Anywhere you want "that sound," but can't justify spending the big bucks for a full-sized Omnia, this little fellow will fit the bill.**
>
> Stephen M. Poole
> CBRE-AMD
> CBNT Crawford Broadcasting Company

what's new! / omnia /

ONE

EFFORTLESSLY SWITCHED STYLES, READY TO MEET YOUR NEEDS.

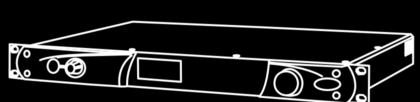
Omnia ONE FM features Omnia's advanced, fully distortion-controlled, pre-emphasized final limiter/clipper, a newly designed digital stereo generator with SCA convenience input, two independently adjustable composite MPX outputs and 19 kHz pilot output for synchronization to external RDS generators.

Rounding out the Omnia ONE AM's processing is Omnia's advanced NRSC compliant, distortion-managed final limiter/clipper, including selectable Low Pass Filter frequencies that support AM HD transmission installations... the same as used in the Omnia ONE's bigger siblings.

Omnia ONE Multicast/DAB features SENSUS audio conditioning technology to minimize codec artifacts as well as restore the fullness and depth that bit-reduction steals. It also features Ultra low-distortion final limiting optimized for the HD codec.

Omnia ONE Studio Pro fulfills the need of a lower delay, full-bandwidth processor for applications that require minimal delay and do not require absolute peak limiting. It is the first studio processor to include a four-band compressor/limiter allowing you precise and accurately defined control while pre-processing music, commercials, remote feeds, or sweetening audio. Applications include recording studios, mastering labs, TV stations, radio headphone feeds... just about any application where signal processing is needed.

Omnia ONE is ready for the future of broadcasting. The flexibility of its platform allows it to stay current as signal processing advancements occur. It uses the Livewire standard for professional networked audio over Ethernet, connecting directly to Axia IP-Audio networks. When used as part of an Axia network, a single CAT-6 cable carries all inputs, outputs and remote control.



Lower delay, full-bandwidth processor for applications that require minimal delay and do not require absolute peak limiting.

OMNIA

ONE

Front View

Back View

GENERAL AUDIO SPECIFICATIONS

	ONE FM	ONE AM	Multicast/DAB & Studio Pro
Frequency Response	± 0.50 dB, 30 Hz to 15 kHz	± 0.50 dB, 30 Hz to 10 kHz	± 0.50 dB, 20 Hz to 20 kHz
Signal-Noise Ratio	> -80 dB de-emphasized, 20 Hz – 15 kHz	> -80 dB de-emphasized, 20 Hz - 10 kHz	> -100 dB, 20 Hz – 20 kHz
System Distortion	< 0.01% THD, 20 Hz – 7.5 kHz. Second harmonic distortion above 7.5 kHz is not audible in the FM system.	< 0.01% THD, 20 Hz – 5 kHz. Second harmonic distortion above 5 kHz removed by system's 10 kHz low pass filter.	< 0.05% THD 20 Hz – 20 kHz
Stereo Separation	> 65 dB, 20 Hz – 15 kHz; 70 dB typical	> 65 dB, 20 Hz – 10 kHz; > 70 dB typical.	> 80 dB, 20 Hz – 20 kHz; 90 dB typical.
Digital Output Level	-24.0 to 0.0 dBFS peak, software adjustable in 0.1 dB steps.	-24.0 to 0.0 dBFS peak, software adjustable in 0.1 dB steps.	-22.0 to 0.0 dBFS peak, software adjustable in 0.1 dB steps.
Crosstalk	> -70 dB, 20 Hz - 15 kHz.	n/a	n/a

A/D Conversion Crystal Semiconductor CS5361, 24 bit 128x over-sampled delta sigma converter with linear-phase anti-aliasing filter. Pre-ADC anti-alias filter, with high-pass filter at <10 Hz.

D/A Conversion Crystal Semiconductor CS4391, 24 bit, 128x oversampled.

Analog Audio Output Left/Right Stereo. Electronically balanced.
Output Impedance 20 Ohms.
Minimum load Impedance 600 Ohms.
Output Level adjustable from -2 dBu to +22 dBu peak in 0.1 dB steps.

External Sync Input External Sync: Allows the output sample rate to be synchronized to an AES3 signal applied to the Ext. Sync input connector. (Does not accept World Clock inputs)

External Sync Range Automatically accepts sample rates between 32 kHz and 96 kHz.

GENERAL SPECIFICATIONS

Analog I/O	2-pair Balanced XLR-type connectors
Stereo Generator Connections (FM style only)	Four standard female BNC
AES/EBU In & External Sync	Shared RJ-45
AES/EBU Out	RJ-45
Ethernet/Livewire	Shared RJ-45
Power Requirements	100-250 VAC, 47-63 Hz., Less than 40 VA
Power Connector	EMI suppressed IEC male. Detachable 3-wire power cords supplied for US and European use.
Power Supply	Internal. Overvoltage and short circuit protected. Meets EN55022, EN55011 Level B Conducted Emissions. EN61000-4-2, -3, -4, -5, -6 level 3 immunity compliant. Full international safety approval. CE marked.
Environmental	Operating Temperature: 32 to 122 deg. F / 0 to 50 deg. C for all operating voltage ranges. Humidity: 0-95% RH, non-condensing.
Dimensions	19" wide x 1.75" high x 16" deep (48.26 cm wide x 13.335 cm high x 40.64 cm deep) 1RU
Shipping Weight	12 lbs./5.5 kg

OMNIA

what's new! /omnia /

A/XE

PROCESSED AUDIO ENCODING FOR WINDOWS

Compelling audio processing and encoding for your audio workstation.

Recapture control of your audio streams with Omnia A/XE - the competition-crushing, market-dominating, accept-no-substitutes Omnia sound for your audio workstation. Sound so pure, clean and compelling that it's almost an unfair advantage.

Omnia A/XE can process audio for a variety of applications, bitrate-reduced and linear. It runs in the background as a Windows service, can be fully-managed and configured remotely with a web browser, and can even process and encode multiple streams in various formats simultaneously.

Encode directly to MP3 or AAC, feed a Shoutcast-style or Windows Media Server in the MP3 format, or stream to Adobe Flash clients through a Wowza Media Server. You can also pair Omnia A/XE with your existing Windows Media, Real, mpgPRO or MP3 streaming encoder.

The new Virtual Patch Cable allows Omnia A/XE to receive, process, and send audio to other software on the PC. Internally encoded Shoutcast or Wowza server streams can be “tagged” with “now-playing” information received from automation systems or another application. We've even built-in a scheduler to allow streams to be started and stopped at specific times, as well as processing presets can be changed on a schedule, perhaps processing the morning show differently than the afternoon one.

Omnia A/XE features adjustable wideband AGC with a three-band compressor/limiter, EQ and low-pass filter, and a precision lookahead final limiter to prevent clipping. Resulting streams are cleaner, clearer, and with more presence and detail.

Hardware Requirements:

Windows XP and later
Minimum 512MB RAM
20MB free hard-drive space
Network Interface Card

OMNIA

what's new! /omnia / more

MORE COOL OMNIA STUFF

Omnia's line of audio processing gear doesn't stop here. We provide a full-spectrum of solutions to fit all of your needs. Below are some of our other popular products.

Omnia-6EX/EXi HD+FM

Omnia-6 is the standard by which all other processors are measured. The clean, pure, crystal-clear sound (bone-shakingly loud, if you want) that's become the choice of #1-rated stations in New York, Los Angeles, Tokyo, Paris, London, Rome, Sydney and Beijing.

Omnia-5EX HD+FM

Omnia-5EX HD+FM contains processing enhancements that result in more bass punch, and more vocal clarity... more exciting and captivating than ever. So powerful, musical, natural, and free of artificial constraints, you'll crave it the first time you hear it.

OMNIA-8X

Combines the power of eight discrete three-band stereo Omnia audio processors in a single, networked box to dramatically improve the sound of your streams.

Omnia-5EX HD+AM

The Omnia-5EX HD+AM is the only processor that accommodates both your analog and digital channels in a single unit that easily integrates with your HD or DRM encoder and transmitter. With smooth, clean processing, it handles AM's programming variety effortlessly, leaving your audience in awe.

Omnia-5EXi HD+AM

Omnia-5EXi HD+AM combines the elegance of the Omnia 5EX HD+AM with Omnia's Diversity Delay features found in the market-leading Omnia 6EXi. Expect more from your AM processing.

Omnia.sg Stereo Generator

Omnia.sg lets you keep the on-air processor at the studio, where it's convenient, while keeping stereo generation at the transmitter, where it belongs. It's the best of both worlds: clear, digital stereo generation and the convenience of locating your processing at the studio.

Omnia-6CD

It was no surprise when friends from the recording industry asked us about developing an Omnia for the demanding requirements of CD mastering studios. Omnia-6CD is a 96 kHz, 24-bit processing platform employing a unique final limiter and precise peak control.

CONSOLE + ROUTER SYSTEM iQ™

AXIA

iQ™

SIMPLY BRILLIANT

EASY INSTALLATION. FAST CONFIGURATION. INTUITIVE OPERATION.

That's iQ, the new console + routing system from Axia Audio. Everything's included: control surface, DSP-based mixing engine, audio I/O, machine-control logic, even an Ethernet switch. There's an available backup power supply, too. Setup couldn't be simpler: connect the iQ control surface to the Core with a single cable, add audio inputs using Cat-5 cable, perform some fast web-based configuration, and your iQ system is ready to broadcast. iQ Simple Networking lets you daisy-chain up to 4 iQ Cores without the need for an external Ethernet switch.

LARGE OR SMALL - THE CHOICE IS YOURS

iQ is the easiest way yet to get started with IP-Audio. Like all Axia systems, iQ is fully customizable and scalable. The 20-input iQ Core comes loaded with I/O for analog and digital audio devices, Mic inputs, an Ethernet switch to connect Livewire devices, and supports large consoles of up to 24 faders. Start with the eight-fader iQ Main Frame; add expansion frames with more faders and capabilities to tailor iQ to your studio's needs. Need more I/O? No problem; just add Axia audio nodes.

FEATURE PACKED

iQ's rich feature set includes 3 dedicated stereo Program buses plus a stereo Utility bus that can be used for phone calls, off-air recording, or as a fourth Program bus. There's automatic mix-minus on every fader, Show Profiles that instantly recall talent's favorite settings, Record Mode one-touch recording, and comprehensive Talkback. Construction details include Axia's hallmark ultra-smooth faders, aircraft-quality switches, and beefy extruded-aluminum frames. High-resolution, ultra-bright OLED displays on the meter bridge can be read instantly from anywhere in the room.

AXIA

SIMPLY BRILLIANT iQ™

iQ MAIN FRAME
Start here to build your custom iQ console + router system. Eight-fader surface with switchable VU/PPM program meters, OLED option displays, Studio and Control Room monitor controls, integrated Talkback system.

8-FADER EXPANSION
Plugs in to the iQ Core to expand your console with eight faders. iQ console frames may be mounted either desktop or drop-in style, as you prefer.

6-FADER EXPANSION
with assignable User Buttons
Six-fader expansion with 10 Film-Cap buttons that work with PathfinderPC software to control routing changes or audio peripherals.

6-FADER TELCO EXPANSION
Six-fader expansion with built-in dial pad and hybrid controls for on-the-board control of the latest Telos telephone hybrids.

iQ CORE
iQ Core is an Axia integrated console engine with 4 mic inputs, 16 Analog inputs, 8 Analog outputs, 2 AES/EBU ins/outs, 8 GPIO, Ethernet switch with 6 100Base-T ports (4 with PoE), 2 Gigabit ports (RJ-45 & SFP). 4 CANBus connectors. Supports consoles of up to 24 faders or three console frames plus accessory panels.

iQ BACKUP POWER SUPPLY
It's always smart to have a backup. For facilities where redundant power is required, the iQ Backup Power Supply delivers complete peace of mind. An easy, single-cable connection to iQ Core automatically provides backup power with automatic switching should the need arise.

what's new! /axia / mixing consoles /

SIMPLY BRILLIANT iQ™

FADER CHANNELS

iQ is simple and intuitive to operate. Options knob gives fast access to source selection, pan and balance and other frequently-used controls. High-resolution OLED display shows source name and status information; context-sensitive Soft key changes function based on your pre-programmed settings.

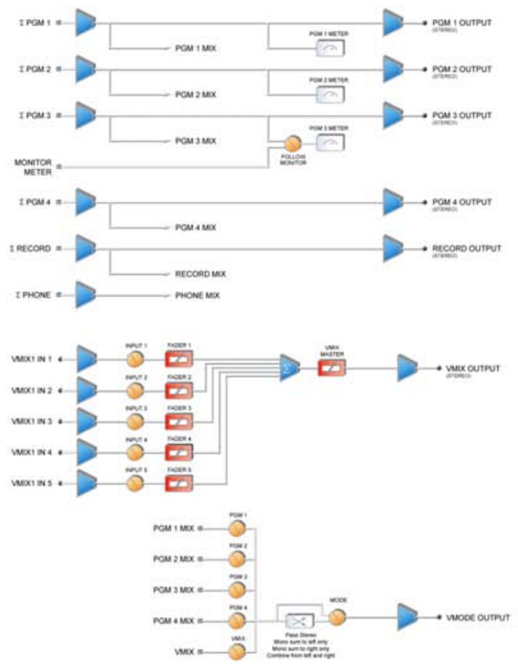
TELCO CONTROLS

Optional Telco Expansion frame includes a built-in 6-line Telos broadcast telephone system. Twin hybrid controls with OLED displays and exclusive Telos Status Symbols give instant caller information; dial pad lets operator dial out directly from the console. Includes Next, Xfer, Block and Hold keys plus Dump key for external profanity delay.

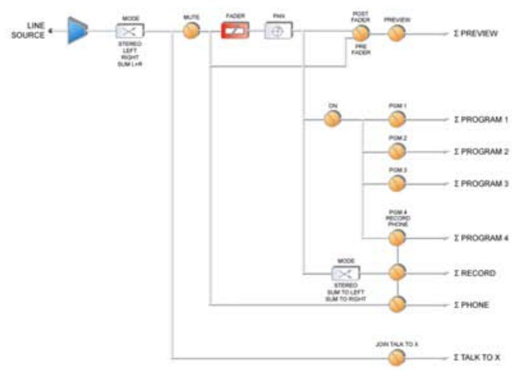
MONITOR SECTION

Monitor choices include all four Program buses plus two external sources. Talk To keys allow operator to communicate with in-studio guests, or to any remote codec or Phone caller using Talk To Previewed function. One-touch Record Mode, Show Profile control, meter options and timer controls complete the package.

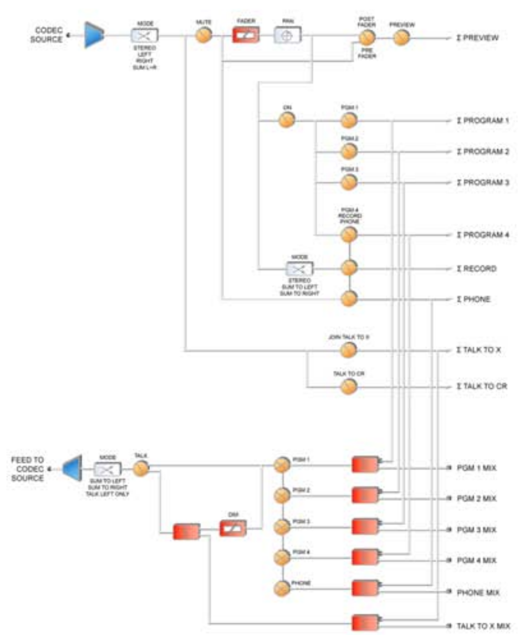
SIMPLY BRILLIANT iQ™



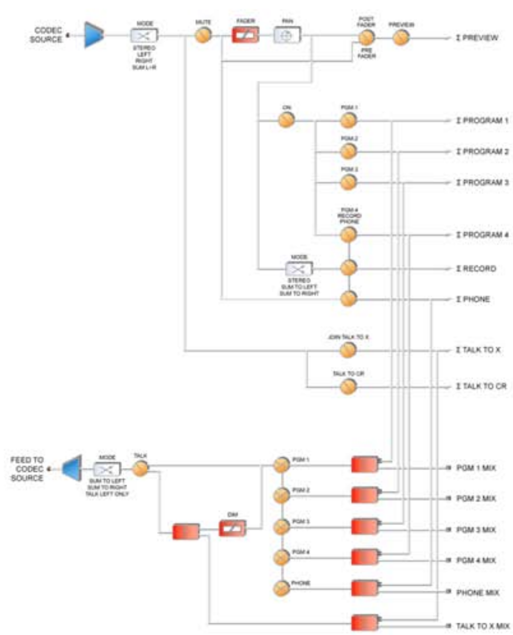
iQ MASTER OUTPUT BLOCK



iQ LINE INPUT BLOCK



iQ CODEC INPUT/OUTPUT BLOCK



iQ PHONE INPUT/OUTPUT BLOCK

AXIA

iQ6

iQ6: MAKES PHONES SMARTER, TOO.

iQ6 FROM TELOS ADDS SIX LINES WITH ONE CLICK.

Up to now, adding phone support to your console could be time-consuming. iQ6 (the new Telco gateway from Telos designed for iQ consoles) makes it simple: one CAT-5 connection and hookup's done.

iQ6 works with the iQ 6-Fader Telco expansion frame to give operators seamless, on-console control of incoming lines and callers. Take calls, dial out, step through pre-screened callers without ever taking your hands or eyes off the board. Exclusive Telos Status Symbols icons let board ops know what's what with just a glance.

You want off-console control? No problem. iQ6 works with the new Telos VX Producer software to give producers more control than ever: take, screen, record and edit calls directly via your producer's PC. There's a built-in soft-phone — just plug a producer's headset into your PC's sound card and you're ready to go. Or use the VX Director phone with its big color screen and animated icons to make short work of pre-screening tasks.

iQ6 accepts POTS or ISDN phone lines, and comes equipped with two 3rd-generation Telos hybrids with Digital Dynamic EQ; the same advanced hybrids found in our Nx phone systems. iQ6 also comes with new AEC Advanced Echo Cancellation from Frauhnhofer Labs, revolutionary technology that eliminates open-mic feedback for smooth, natural hands-free conversations.

Naturally, since it's part of the iQ system, iQ6 saves you money and time. It plugs right into your Livewire network, eliminating the cost of discrete I/O, cabling and soldered connectors, and drastically reduces installation time. One skinny CAT-5 cable carries six lines' worth of audio, hybrid control and backfeed.

POWERSTATION, THE ADVANCED, INTEGRATED CONSOLE ENGINE THAT'S BUILT LIKE A TANK

UNPACK IT, INSTALL IT… FORGET IT.

Meet PowerStation: the latest console advancement from Axia, the company that invented IP-Audio. From the beginning, IP-Audio networks saved broadcasters time and money by providing them with an easy way to bring together all the separate pieces of the broadcast studio – audio devices, consoles, machine control, delivery systems, routing switchers – and combine them into one flexible, easily-managed system.

PowerStation makes IP-Audio even easier to deploy by combining six separate devices — a DSP mixing engine, a console CPU and power supply, audio I/O, GPIO and an Ethernet switch – into one unit, a self-contained console engine that's engineered – make that over-engineered – to ensure years of reliable, trouble-free service.

Our clients have always raved about the time and money saved by an Axia network. PowerStation helps save even more, dramatically streamlining studio builds: put PowerStation in your rack, connect your Axia console with just a single cable, plug in your audio devices and name your sources with the user-friendly web interface. Add power and you're on the air. It's that simple!

THE INDESTRUCTIBLE CONSOLE ENGINE

FACTS AND FEATURES

PowerStation Main is where you start, with half-a-dozen essential tools combined into one compact unit. Inside you'll find a bulletproof mixing engine capable of handling consoles up to 40 faders, a massive power supply (with optional redundant power), machine control ports, and audio I/O, all in one box. How much audio I/O? You'll find two Mic inputs, four Analog inputs and six outputs, two AES/EBU inputs and two outputs, four GPIO ports, each with five inputs and five outputs. And there are 16 built-in Ethernet ports with Livewire for single-cable connection of Telos phone systems, Omnia audio processors and other Axia equipment, as well as gear from an ever-expanding list of Livewire partners. And two Gigabit ports with SFP let you connect to copper or fiber networks.

If you need more I/O, simply connect a PowerStation Aux to instantly double your Mic, Analog, AES and GPIO ports. PowerStation Aux connects quickly and easily, with just one Ethernet cable and one power connector.

Speaking of power, I/O isn't the only thing that doubles when you connect a PowerStation Aux: you also get a redundant, backup power supply with built-in switchover. Most redundant supplies protect only the console, but since PowerStation is an integrated console engine, the mixing engine, audio I/O and network switch are protected as well — a feature you'll not find anywhere else.

OVER-ENGINEERED TO AXIA STANDARDS

You want your console to be more than just reliable — you want it built like a battleship. You want the absolute peace of mind that comes from knowing your gear will never let you down. So our team of obsessive console designers made sure that every part of PowerStation was chosen for its ability to give constant, uninterrupted service, 24 hours a day, 7 days a week, 365 days a year. There are no compromises: PowerStation uses only best-of-the-best components, like studio-grade mic preamps and 24-bit, 256x oversampling A/D converters, a rigid, EM-tight chassis, an ultra-reliable DSP platform (not a common PC motherboard) and a hardened power supply designed for unfailing service, even in the harshest environments.

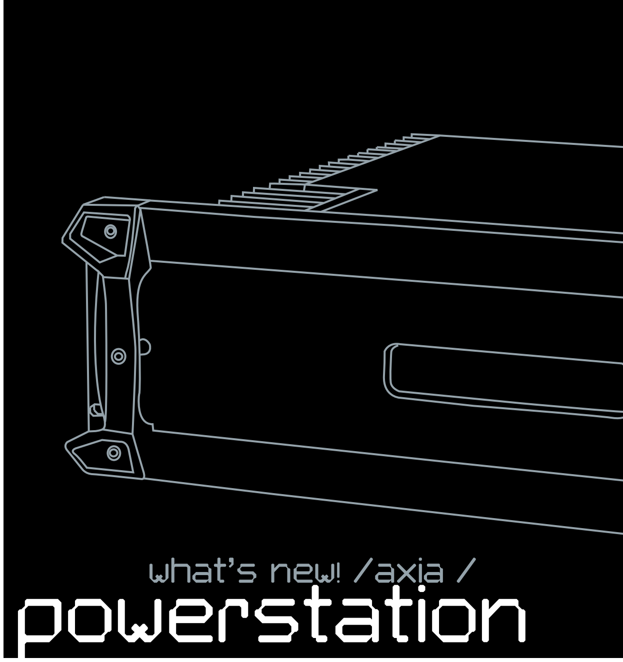
AXIA

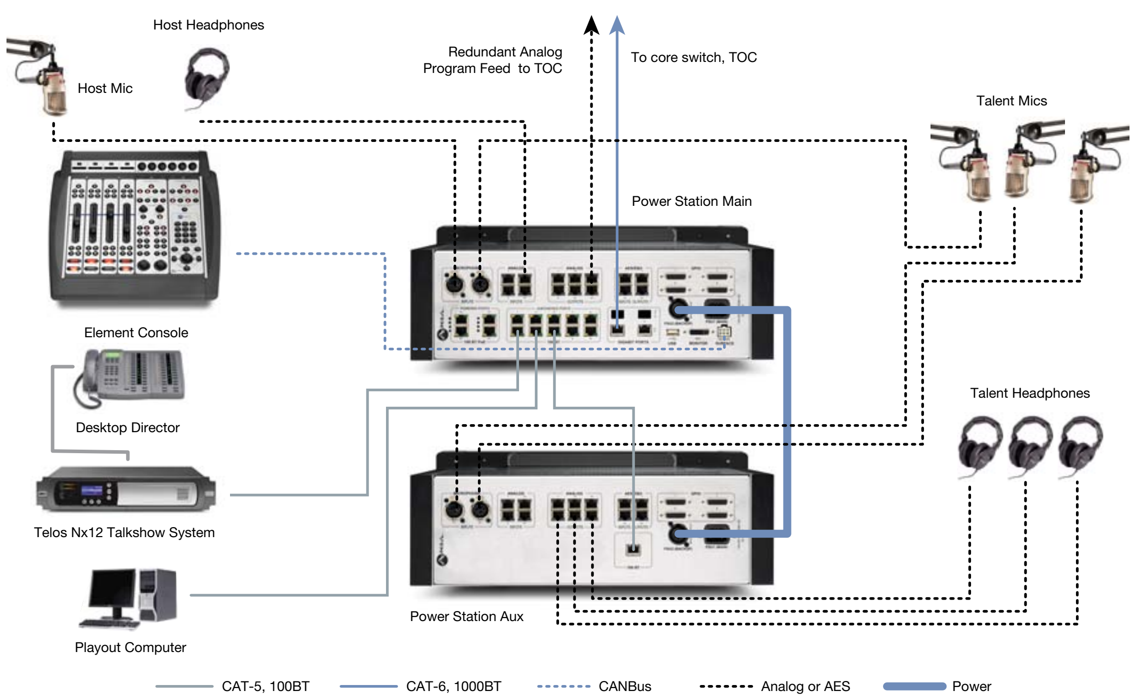
THE STANDALONE NETWORK

We designed PowerStation to be the world's first networked broadcast console that doesn't need a network. It's completely self-contained: it works flawlessly as part of a large network, but if you unplug its network cable, it's completely unaffected. Think of it as an "island of reliability."

Not only does PowerStation make it easy to build stand-alone, independent studios, it also makes it easier than ever to network them together. An exclusive new Axia feature we call Simple Networking allows up to 4 PowerStations to daisy-chain without the need for a separate switch (although you can add one to build even larger networks).

Build just one studio, or a dozen, at any pace you choose — your PowerStation network is ready to expand when you are, up to as many as 10,000 stereo channels.Want to know more about PowerStation? Visit AxiaAudio.com/PowerStation/ for the whole skinny.





mixing consoles

ELEMENT 2.0

POWERSTATION WORKS WITH THE AXIA ELEMENT BROADCAST MIXING CONSOLE

With more than 1,500 consoles already on the air, Element is a huge hit. And now, thanks to suggestions from our clients, it's better than ever. Element 2.0 has cool features like Omnia headphone processing presets to give talent that "air sound", super-accurate metering with both peak and average displays, one-touch phone recording with automatic split-channel feed, automatic mix-minus for every fader, an eight-channel Virtual Mixer that lets you combine multiple audio streams and control them with a single fader, and metallic bronze or silver module overlays. And we haven't even begun to tell you about Element's Show Profiles that instantly recall talent's favorite settings, its built-in Telco controls, fully-integrated talkback/IFB and Mic processing by Omnia.

And durable? Element is nearly indestructible, ready to take whatever pounding ham-fisted jocks dish out and keep going. You want examples? Element's avionics-grade switches are rated for more than two million operations. What look like ordinary rotary controls are, in reality, bullet-proof optical encoders — no wipers to wear out or get noisy. The silky-smooth conductive-plastic faders actuate from the side, not the top, so dirt and grunge stay out. The high-impact Lexan module overlays have their color and printing applied on the back, where it can't wear or chip off. The frame is made from thick aluminum extrusions that are stronger than truck-stop coffee.

To find out even more about Element, visit AxiaAudio.com/Element/ . Grab some coffee and prep for a good, long read -- remember, our marketers get paid by the word.

ip-intercom

(GO AHEAD, TALK AMONGST YOURSELVES)

NOW HEAR THIS

If you can hear it, you should be able to record it, edit it, or get it on the air. If you can talk to it, you should be able to cue it and feed it mix-minus or IFB. Anything else is tin cans on a string. IP-Intercom puts no barriers between your broadcast audio and your communications channels. Unlimited full-bandwidth access to any studio, news or sports venue, office, hallway, broom closet or latrine — if that's what you're into. Talk and listen to individuals or groups hands-free, with no echo or feedback.

GAB ON. GAB OFF.

We believe only Hula dancers should need their hands to talk. Broadcast communications ought to be natural and hands-free. That's why the Axia Intercom incorporates Advanced Echo Cancellation by Fraunhofer Labs. It literally eliminates open-mic feedback without speaker muting. Just open a channel and start talking. You can use your hands for more important things, like endorsing checks, signing autographs, or Wii bowling.

BUZZ OFF

The last thing you need during a breaking story or transmitter failure is hum and buzz getting between you and the guy you need to talk to. Like all Axia gear, our intercom system is completely digital inside and out. Other systems try to make you think they're digital by piping their analog signals over CAT-5 cables, but we think that's a bit like putting an abacus app on an iPhone.

AXIA

ip-intercom

INTERCOMS EVERYWHERE

Axia SoftCom software allows anyone with a networked computer to have a virtual intercom station. Just think: you'll never again lose track of who's where when you need them to get on-air or get you coffee.

WHERE THERE'S A WIRE

The advantages of IP and Ethernet – low cost, easy installation and maintenance, efficient infrastructure – are a given. IP links everything, and now this includes your intercom system. Installation is a simple single-click connection. And you can expand it like breeding rabbits. Plug as many stations into your switch as you want and add on from there. Then start talking. And if you move to a new location, no biggie — just pick up the gear and take it with you. IP-Intercom is portable so there's no expensive hard-wired custom-cable multi-pair infrastructure mess to deal with.

PLAYS WELL WITH OTHERS

Don't have an Axia system? That's OK. You can still save money, increase efficiency, and decrease the hard-wired infrastructure hassle by choosing IP-Intercom. It's a stand-alone system with I/O that will accommodate multiple consoles. But if you do have an Axia system, you'll get more operational goodies like seamless console integration with these nifty drop-in modules. We're just sayin'.

TOUCH THAT BUTTON

So you've gotta be a genius to use it, right? Actually, any acne-challenged intern with an index finger can operate this system with ease. The web interface makes setup simple. Sharp, high-contrast OLED displays are easy to read from anywhere in the room. And our clever callback feature makes sure you'll never miss a call, no matter what you're doing. Come to think of it, that intern doesn't even need a finger.

FAMILY TIES

In the world of Axia audio products, think of IP-Intercom as the talkative little brother. It's an integral part of the family, so of course it links with existing Axia networks and consoles. Just connect to the Ethernet with one RJ-45 cable and the intercom is ready to play. In fact, the intercom audio is ready to go directly to air. Or you can feed IFB board audio to intercom callers. The possibilities are endless.

WE HEAR YOU

As always, Axia has free 24/7 technical support, 365 days a year (the loneliest support team on earth). And our warranty is the most comprehensive in the industry – 5 years parts and service. (Really, you should call the technical support guys once in awhile just to say hi.)

MATRIX: TERMINATED

Imagine a digital intercom system with no central matrix. Actually, don't bother. We've already built one that saves on cost, installation time, special plug-in cards, and space. It's real plug and play that works every time — even when you need to add a station, or reconfigure the ones you've got.

what's new! /axia /

network

THE AXIA AUDIO NETWORK.

REPLACE THE ROUTER

Networking revolutionized computing. Now it's doing the same for broadcasting! Before the network, building a complex facility with lots of I/O to send to multiple destinations required a router, a big, complex piece of proprietary machinery. But now there's no need for an expensive router — Axia makes it simple and economical to design a routing switcher custom-tailored to your specific needs. Build a network as tiny as 8x8 or as large as 4096x4096… or more.

Broadcasters love Axia's modularity and scalability — there are already over 1,500 Axia Livewire networks on-the-air. Combine Audio Nodes, Router Control Panels, an Ethernet switch and our sophisticated PathfinderPC or PathfinderPRO Router Control software; configure using any standard web browser, and you're ready to go.

Getting your audio into the network couldn't be easier: just connect analog and AES/EBU sources to Axia Audio Nodes. Getting audio from PC workstations is even simpler: use the IP-Audio Driver for Windows, Mac or Linux to send and receive content from audio workstations and playout systems directly to and from the network — without the expense of soundcards or audio converters. There are also plenty of broadcast products from Axia Partners that have Livewire built-in — a single CAT-5 cable carries audio I/O, start/stop logic, clean feeds and more.

BUILDING BLOCKS

It's easy to build an Axia network customized to your exact needs (but with the ability to easily add capacity). There are a wide variety of Audio Nodes and Router Control Panels that can be conveniently placed wherever your audio devices are located.

The Analog Line Node has eight balanced stereo inputs and eight balanced stereo outputs, all on RJ-45 connectors, and employs premium-quality A/D/A converters and low-noise components to provide superior audio performance for high-end studio use. Each input can accommodate either consumer-level -10 dBv or professional level +4 dBu equipment.

The AES/EBU Node provides eight stereo AES3 inputs and eight AES3 outputs. Sample-rate conversion is available on all inputs; an input can also be used to sync the Livewire network to a house clock. Like all Audio nodes, the AES Node features front-panel confidence metering for each of its inputs and outputs.

The Microphone Node has eight studio-grade mic preamps with selectable Phantom power and software-adjustable gain. There are also eight balanced stereo analog line outputs to conveniently deliver headphone and studio monitor feeds back to the talent. Inputs use XLR connectors; outputs are on easy-to-install RJ-45s.

AXIA

PathfinderPC and PathfinderPRO Router Control Software enable creation of sophisticated custom routing applications — from simple point-to-point routes to multi-point scene changes that can reconfigure an entire facility manually, at a specified time, or in response to an event trigger. Pathfinder automatically scans the network, gathering information about audio sources and destinations. Administrators can then “point and click” to send audio from one place to another.

With Pathfinder, you can combine audio and machine logic into a single “virtual router”, “watchdog” audio sources with metering and Silence Sense, and make custom applications with a graphical “Stacking Events” editor that eliminates script writing.

But the Pathfinder feature our clients rave most about is its ability to monitor audio sources and take action if there’s a problem, such as automatically switching to backup sources, flipping GPOs, and even sending an email to your Blackberry or iPhone that says “All’s well. I’ve made some changes.”

Pathfinder comes in single-server (PathfinderPC) and dual-server enterprise (PathfinderPRO) versions.

NEW X1 Router Control Panel lets you instantly route any source on your Livewire network to a specified destination. Scroll through available channels using the bright, high-visibility display; then just press the “Take” key to make your route change. 1RU rackmount package for convenient placement anywhere control is needed.

NEW X2 Router Control Panel provides the routing functionality of two X1 Panels in a single chassis; each separate control routes any available network source to an individually specified destination.

NEW XY Router Control Panel is the ultimate routing tool: select any source on the tack-sharp OLED display and send it to any destination with the click of a button. 1RU rackmount package for convenient placement anywhere control is needed; perfect for TOC, newsroom or head-end operations.

THE AXIA AUDIO NETWORK.

REPLACE THE ROUTER

NEW 8-Button OLED SoftSwitch provides high-visibility Router Control from any studio turret or equipment rack. Its eight bright, sharp OLED (Organic Light-Emitting Diode) readouts can display anything from simple text to color photos, and can be seen from nearly any angle — across the table, or across the room. Use PathfinderPC to program custom routing commands you can invoke instantly with the touch of a button.

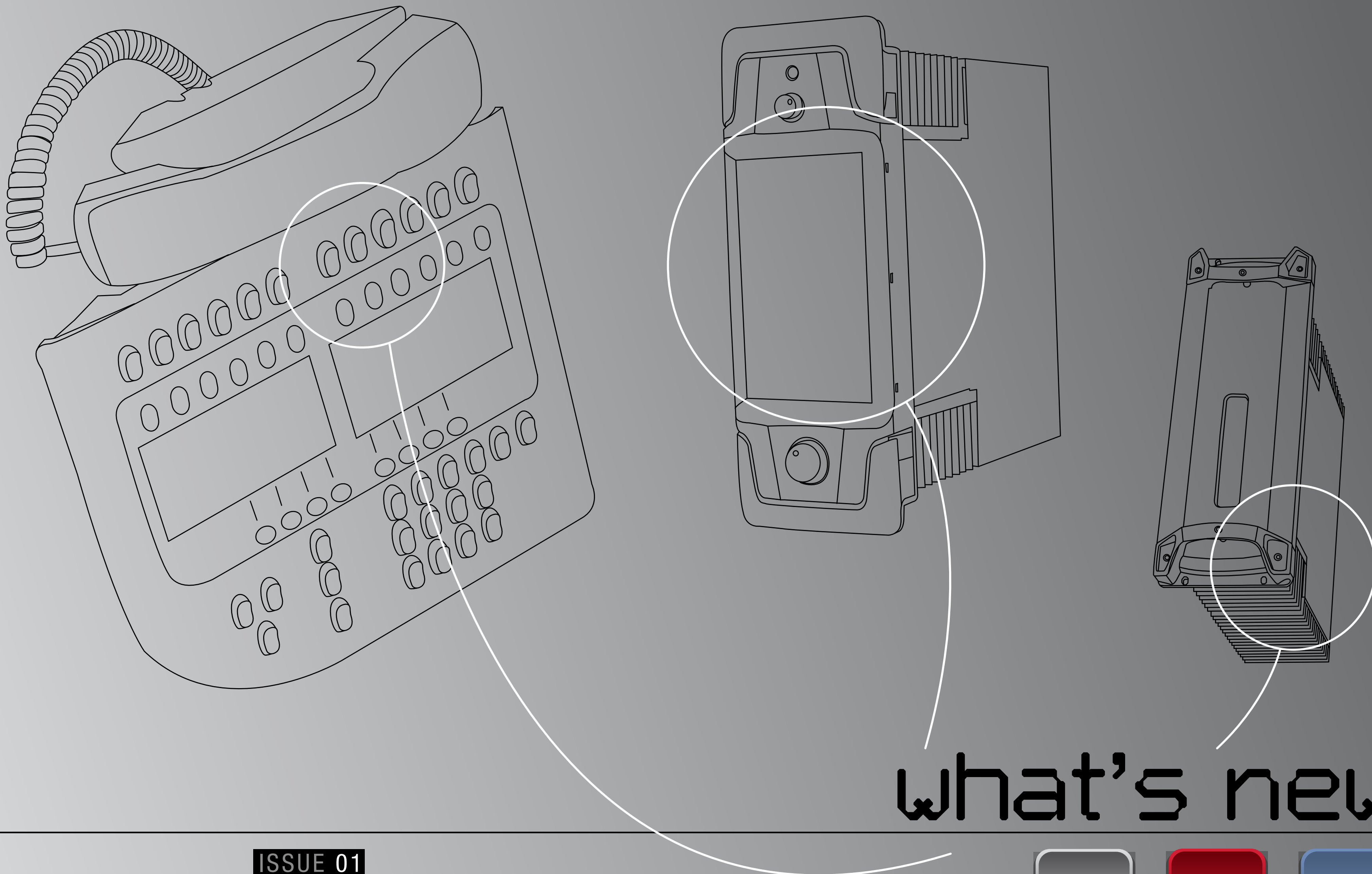
SmartSwitch Router Control Panels feature backlit LCD buttons with dynamic text and color to provide one-touch remote access to often-used machine-control or software functions. Multiple pages of button assignments can be programmed and recalled with just a touch; use PathfinderPC's Stacking Events Editor to map single commands or complex routing salvos to any button. Available in 9- and 17-button rackmount models, 4-button guest accessory panels and 5- and 10-button modules for Element consoles.

Use these Film-Cap Router Control Panels when dynamic-text capabilities are not required; lighted aircraft-grade switches provide fast execution of router salvos, machine-control or software functions programmed using PathfinderPC Router Control software. 5-, 10- and 15-button rackmount models are perfect for use in a studio turret, TOC control panel or equipment rack. A 5-button guest accessory panel and 5- and 10-button Element console modules are also available. Place film labels under the clear button caps; set the LED backlights to any of 8 different colors.

The Router Selector Node combines the routing functions of a Router Control panel with the audio outputs of an Audio Node. In addition to analog, AES3 and headphone outputs, the Router Selector Node also features an analog and an AES3 input — ideal for production or news studios where operators both create and play audio streams. Eight convenient "radio buttons" can be quickly programmed for instant access to favorite sources.

The GPIO Node provides 8 logic ports for machine control, each with 5 opto-isolated inputs and 5 opto-isolated outputs. A logic port can be associated with any input or output; logic is routed along with associated audio. Logic commands can be triggered by console on/off buttons, or automatically via custom event commands programmed with PathfinderPC Router Control software.

AXIA



what's new

ISSUE 01

TELOS OMNIA AXIA

CATALOG SUPPLEMENT 2010

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